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PHYSICAL ACTIVITY

The Superbooth show in Berlin has become one of the highlights of the electronic musician's calendar, and for good reason. Who could fail to enjoy an event that allows you to check out the latest innovations in synth design, immerse yourself in ambient drone music and eat currywurst, whilst exploring an East German children's centre? In a forest?

The buzz products at Superbooth are a good indicator of industry trends, and the direction of travel this year was intriguing. There were, naturally, plenty of new analogue synths and modules, with significant launches including UDO's droolworthy Super Gemini, Tiptop Audio's ambitious polyphonic Eurorack standard, and a parting gift from Wasp and OSCAR designer Chris Huggett in the shape of the PWM Mantis. But there was also an unprecedented focus on a previously neglected synthesis method.

Physical modelling was pioneered by Yamaha nearly 30 years ago, but although the technology has since delivered some stunning software instruments recreating pianos and orchestral instruments, it has never really broken into the mainstream in the way that, say, wavetable synthesis has. If the stand-out product launches at Superbooth 2023 were anything to go by, that may be about to change.

With their Objekt soft synth, Reason Studios seem to have reined in the tendency of physical modelling to be unpredictable and clangorous. There's something uncanny about its output, as though we're hearing real instruments from an alternate universe. By contrast, Erica Synths' Steampipe positively revels in the tendency of physical modelling to be unpredictable and clangorous, yet somehow retains an organic, musical quality.

Most interesting of all was the prototype "acoustic synthesizer" that Korg Berlin were showing in their wagon in the woods. (Superbooth is the kind of event where multinational corporations show things in wagons in the woods.) This is not physical modelling synthesis but physical synthesis, using electrostatic pickups both to capture and to modify the resonant behaviour of steel tongues. There is clearly a fair way to go before it can form the basis of an actual product, but the concept and the execution are fascinating. A personal highlight of the show was the opportunity to interview designer Tatsuya Takahashi for the *SOS* YouTube channel: https://sosm.ag/superbooth-takahashi.

Finally, talk of shows brings me to our own GearFest UK, which was such a success in 2022 that it's grown into a two-day event this year. Across the weekend of July 15th-16th, GearFest will give you the chance to get hands-on with the latest pro audio and synth gear at Tileyard in King's Cross, Europe's largest studio complex. Visit www.gearfestuk.com to register for tickets and check out the ever-growing list of exhibitors, and we'll see you there! ■■■

Sam Inglis
Editor In Chief

> "The buzz products at Superbooth are a good indicator of industry trends, and the direction of travel this year was intriguing."

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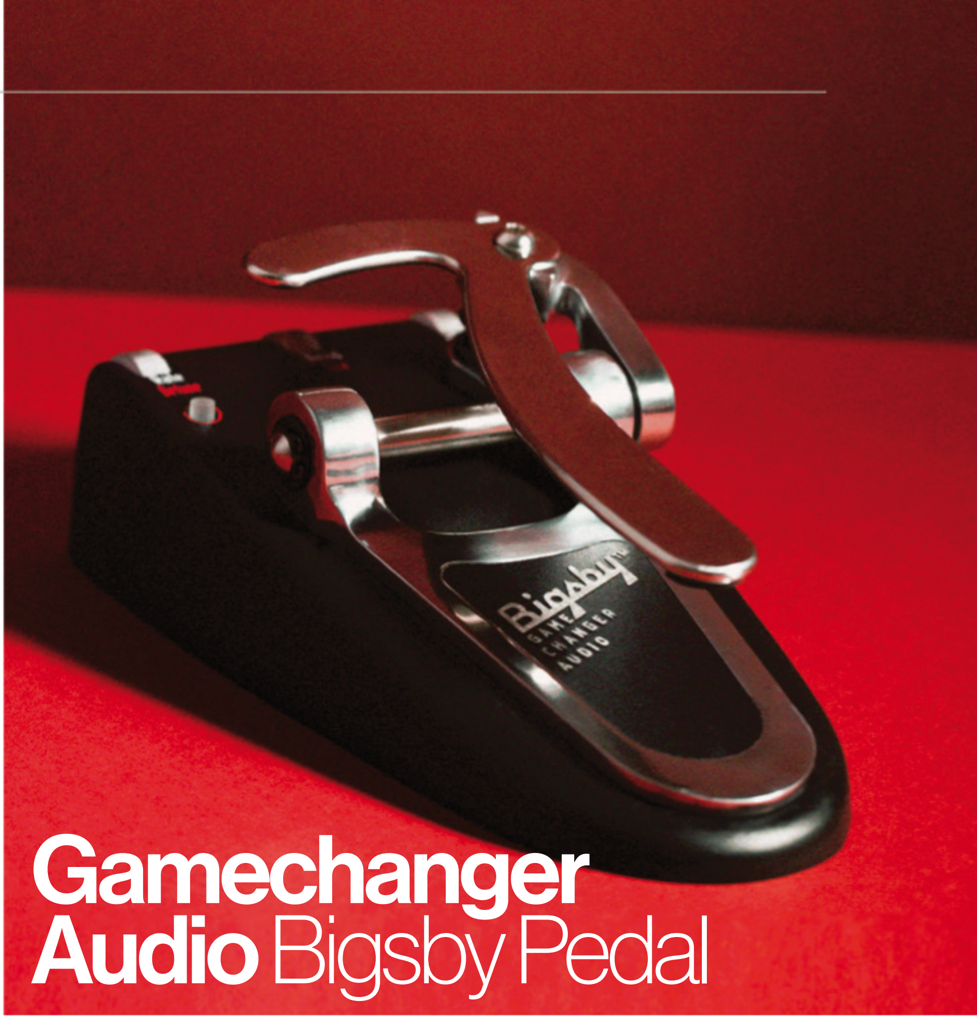

Gamechanger Audio Bigsby Pedal

Polyphonic Pitch-shifter

There's more to this stylish-looking pedal than meets the eye.

PAUL WHITE

Designed in collaboration with Fender, Gamechanger Audio's Bigsby Pedal is a polyphonic pitch-shifter that aims to achieve smooth polyphonic real-time pitch-shifting, while preserving the natural sound of electric and acoustic stringed instruments. Smooth pitch-shifting is notoriously difficult to achieve, especially on polyphonic sources, and the shifting algorithm in this pedal, which the company describe as a 'hybrid' and which runs on a SHARC DSP chip, does sound smoothest with smaller pitch changes — but it also sounds surprisingly good when you go for the full octave, with far less in the way of noticeable artefacts than other polyphonic pitch pedals I've tried.

Overview

The hardware is beautifully engineered, a specially designed arm allowing foot control in sharp and flat directions. There's no bypass button; the pedal is always active, but a light on the top is off when the pedal is in its centre 'no shift' position, and glows brighter as you move from the centre, and the transition from dry to the pitch-bent positions sounds seamless. An optical system tracks the pedal movement and a strong spring returns it to the mid position when no pressure is applied.

Three dual-function, rotary wheel-style controls with internal illumination address Rate Blend and downward pitch range (Depth) when their backlights show white. When the backlights are red, the controls address Detune, Tone and the upward shift range. A small button on top of the case switches modes from white to red, and a switch on the rear flips the direction in which the pedal operates, as well as being used in a routine to lock the rotary controls to prevent inadvertent adjustments.

In addition to providing controlled pitch shifts of ±1 octave, in red mode the Detune parameter helps to emulate the mechanical imperfections of a traditional guitar vibrato arm, whereby the strings' pitches don't all move by exactly the same interval, resulting in a slight-chorus-like detune. When in Rate mode (white lights) the same control applies an LFO-controlled pitch modulation that gets progressively deeper at more extreme pedal positions, its speed determined by the Rate/Detune knob position. Rate goes from a very slow undulation up to somewhere north of a fast Leslie speaker, and if you want whammy-type fifths or octaves you'll probably want Detune and Rate turned off. Blend/Tone mixes in some of the dry sound, producing a chorus-like effect that gets more intense as more pitch-shift is applied, while in red light mode it sets the tonality of the shifted signal, operating as a type of tilt EQ.

A quarter-inch TRS jack allows the pedal to be used as a sprung-to-centre expression controller, with adjustable depth, rate and offset parameters, and this can be used while you're pitch shifting. Another jack accepts momentary or latching footswitches, which activate a Latch mode: a constant pitch interval is set using the Depth knob, and the pedal can make pitch changes of up to two semitones. A TRS mini-jack MIDI input also makes it possible to control all the pedal's parameters via CC messages. MIDI also facilitates the creating and saving of 10 presets and sync'ing the Rate's LFO speed to subdivisions of MIDI tempo via MIDI Clock. A MIDI data chart is included, and all the settings are done using the existing controls, via the usual 'hold this, press that' kind of sequence. There are also details on how to get the pedal to recalibrate if anything goes wrong. In normal use, the pedal performs an auto-calibration routine each time it powers up (so it's important not to move the arm from its centre position when switching on; otherwise there will be a calibration error!)

Pitch Perfect?

I didn't quite know what to expect when I first heard about the Bigsby pedal but having had my hands on one, I have to say that I'm pleasantly surprised, both by its physical construction and by its sound quality. It produces smooth pitch transitions that are at least the equal of anything else I've tried — and better than most. The physical aspect takes some getting used to. It's tricky to rest your foot on the pedal between periods of pitch shifting, as it's too easy to move it one way or the other, resulting in unwanted shifts. Also, as the pedal is longer behind the pivot point than in front of it, it's easier to control shifts that require the long 'arm' to be pushed down than the short one. Once you get used to the mechanics, though, it all works extremely well, and it also offers much more 'under the hood' functionality than I'd expected too. ▮▮▮

summary

Cool and quirky, this polyphonic pitch-shifter offers comprehensive MIDI support — and sounds great!

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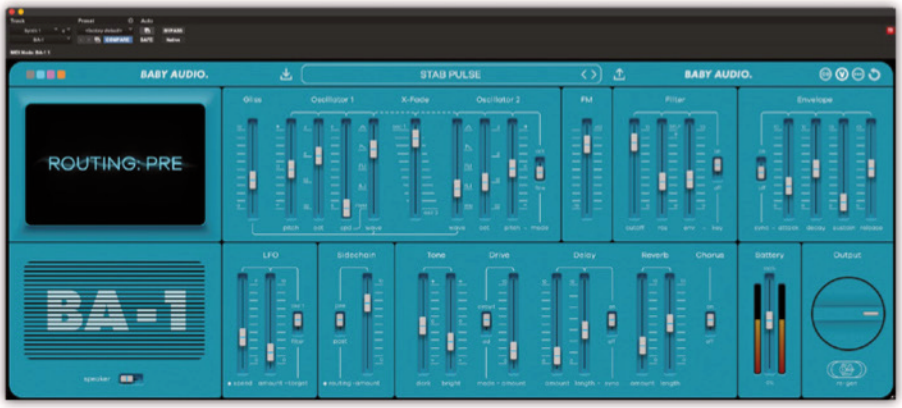
Baby Audio BA-1

Software Synthesizer

Baby Audio's first soft synth is inspired by the baby of the Yamaha CS range, the small but immediate CS-01.



SAM INGLIS

When plug-in developers emulate vintage synths, there's a tendency to focus on the big beasts. And that's perfectly understandable: given the choice between a Memorymoog and a Micromoog, most of us wouldn't have to think twice. However, that's not to say that smaller and cheaper instruments deserve to be overlooked. Despite, or possibly because of their limitations, they often have a distinctive character and sound of their own.

The Yamaha CS-01 is a case in point. The baby of the range that also included the mighty CS-80 and the versatile CS-30, it was a cheap and cheerful portable monosynth with a basic feature set. But it had a distinctive charm nonetheless, thanks in part to its lo-fi built-in speaker and ability to run on batteries. So enamoured are Baby Audio of this miniature marvel, in fact, that they've taken it as the inspiration for their first soft synth.

Hit Me Baby One More Time

BA-1 is more than just a pedantic recreation of the CS-01, though. Baby Audio have added numerous additional features, including a second oscillator with frequency modulation, a 'side-chain' module and an effects section with Tone, Drive, Delay, Reverb and Chorus. Unlike the original, BA-1 can be played polyphonically, and there are many more minor improvements such as the fully variable resonance control and optional key tracking in the filter section. But what is most apparent is Baby Audio's determination to capture the quirks and unique features of the original. Not only have they emulated the loudspeaker, with its characteristic nasal, buzzy sound, but they've also added a slider that allows you to imitate the effect of running a CS-01 on dying batteries.

Even with all Baby Audio's improvements, BA-1 is still a relatively basic instrument. It has only a single LFO and envelope generator, and no real routing flexibility. In fact, there are probably more versatile synths than BA-1 bundled with your DAW of choice — but that's not really the point.

What BA-1 has going for it is immediacy. With such a compact user interface, it's easy to take in all the settings at a glance, and equally simple to figure out what you might need to change. Said interface can be freely resized, and comes in your choice of four entertainingly retro colours. And although there aren't all that many controls, every single one seems to do something worthwhile throughout the entire length of its travel. It's very hard to make this synth sound bad, a point which is brought home when you experiment with the 're-gen' random patch generator. In 10 minutes of stabbing at that one button, I generated half a dozen sounds so good I felt compelled to save them as presets.

Babyface

In terms of character, BA-1 is punchy and in your face, but not abrasive or edgy. It's the synthesizer equivalent of a small guitar amp like a Fender Champ: not super-big or hi-fi, but what it does fits perfectly in the track, sitting naturally at the front of the mix without any effort on the user's part. The second oscillator is a very handy addition, and the substitution of white noise for PWM here opens up a lot of new sonic territory. I also particularly liked the FM, which is unexpectedly musical. Instead of atonal metallic clonks, it usually generates complex, rich-sounding but harmonious results. Like the original, BA-1 provides glissando instead of portamento, which can be a lot of fun.

Drag the Battery slider below halfway and you'll begin to hear pitch instability on sustained notes, with noise and distortion creeping in at the lowest settings. It probably isn't something you'd use all the time, but I can imagine automating the slider to bring it in for effect.

The effects are well chosen, and sound as good as you'd expect from a plug-in developer who previously specialised in good-sounding effects. The side-chain option introduces something resembling a tempo-sync'ed tremolo, which can be quite cool, though I'd have welcomed a speed multiplier. And the Tone and Drive sections somehow provide fairly extensive control over the colour of the output tone without losing its character.

If you're looking for an all-singing, all-dancing soft synth that offers endless depth and tweakability, you won't find it here. But if what you're after is a cheap, straightforward and charming virtual analogue synth that will put a smile on your face and deliver instant gratification, BA-1 is hard to beat.

summary

With a characterful sound and affordable price tag, Baby Audio's BA-1 is a worthy tribute to the plucky Yamaha CS-01.

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Rode Wireless ME

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Recording podcasts, interviews and video content couldn't be easier with Rode's compact new wireless mics.

PAUL WHITE

Rode's Wireless ME is an ultra-compact wireless microphone system designed to meet the needs of content creators. The transmitter and receiver are both very small (44 x 45.3 x 18.3mm), and weigh just 32g each, yet they both have built-in microphone capsules, so they can be used without having to plug in external mics. However, there is also a mini-TRS jack connection on the transmitter for an external lavalier or camera mic, should one be needed.

Having a second mic located in the receiver is invaluable when recording a conversation between a third party and the operator, while the compact size is a bonus when working with video. The mics themselves use omnidirectional back-electret capsules, and conversion is 24-bit at up to 48kHz, with a frequency range of 20Hz to 20kHz (no limits specified), a dynamic range of 100dB and a maximum SPL of 122dB SPL. An equivalent noise figure of 22dBA is specified, which is more than adequate for the vast majority of situations.

The system incorporates Rode's GainAssist technology, which keeps recording levels in check automatically. This uses what Rode describe as 'intelligent' algorithms to automatically control audio levels and to prevent clipping. In Auto GainAssist mode, the system aims to achieve a consistent sound level, which is useful when the audio source is moving around. Dynamic GainAssist mode captures more of the natural dynamics of the source while still trying to stabilise the overall level. You can also turn off GainAssist if you prefer to set the gain manually.

Utilising Rode's Series IV 2.4GHz digital transmission with 128-bit encryption, the Wireless ME boasts a range of over 100 metres depending on the environment, and requires no licence to operate. The integral rechargeable lithium batteries allow an operating time of up to seven hours, and can be recharged using any 5V USB port or charger.

Two furry windshields and a storage pouch are included, along with four cables enabling the receiver to be plugged directly into cameras, computers, iPhones (iOS 16 or later) and Android (9.0 or later) smartphones. The system is fully compatible with Rode's Capture, Connect and Reporter apps. Additionally, it is possible to add another transmitter in situations where three mics are needed. The Rode Connect and Rode Central apps provide controls to enable or disable the mic in the receiver, switch the GainAssist mode to Auto, Dynamic or off, and to pair or unpair up to two transmitters. You can also switch between Split and Merged recording modes (the two mic signals are mixed or kept separate), adjust the output gain, change the LED brightness and update the firmware. Changes made using the apps are remembered when the device is disconnected. The receiver's connectivity LED shows green if its mic is on or orange if it is off.

Podcasters will find Rode Connect especially useful, as it is capable of connecting up to four Wireless ME devices to one computer, and also includes a section of pads for launching sound effects.

Knowing ME, Knowing You

As shipped, the transmitter and receiver are already paired, though re-pairing is very straightforward. The transmitter has an LED to show charging/battery condition and another to indicate that it is paired with the receiver. The battery LED flashes green while charging and then remains steady when charging is complete. A power button is located on the bottom edge of the case and the integral mic grille is at the top, with the mini-TRS mic jack alongside. A USB-C port is located on one side and an integral sprung clip allows fixing to clothing.

The receiver looks very similar, but rather than having an external mic input, it has a mini-jack headphone output alongside its USB-C port. Setting up proved easy enough and the audio quality is impressive given the small size of the devices, with a natural-sounding clarity and minimal noise. The range is generous in line-of-sight situations but I began to get dropouts as I moved further away and into different rooms separated by brick walls. However, in most of the applications I can envisage, the transmitter won't be too far from the receiver or separated by solid walls.

Connected via USB, the receiver shows up as an audio interface in a typical DAW, though recording can be handled quite effectively within the Rode Connect software too, so unless you need to add specific effects to the recording there's no need to involve a DAW. As an affordable but high-quality wireless solution for creative work where you don't want to get bogged down by unnecessary complications, the Wireless ME system gets a thumbs up.

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Black Lion Audio Bluey 500

Mono FET Compressor

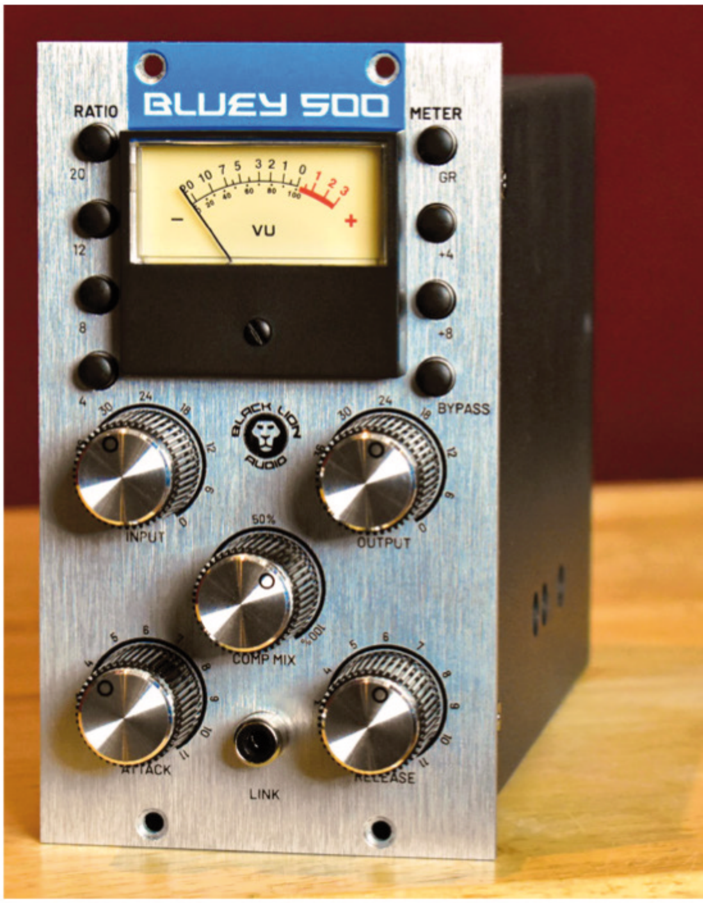
BLA have packed their take on the classic 'blue-stripe' 1176 into a 500-series module.

NEIL ROGERS

Black Lion Audio released their 19-inch rackmount Bluey compressor back in 2020, and in my review (www.soundonsound.com/reviews/black-lion-audio-bluey) I found much to like. You will find more detailed background in that review but, essentially, it is Black Lion's take on Chris Lord-Alge's (CLA) favourite 'blue-stripe' 1176 compressor, a unit which they were invited to investigate, and which had become heavily modified over years of servicing by different technicians. Black Lion's engineers found that this device exhibited a more mid-forward sound than a 'normal' 1176, and that this characteristic often seems to work especially well for presenting vocals in the context of a mix. In collaboration with CLA, Black Lion carefully measured and then matched the components and characteristics of this unique device, so they could give every engineer a chance to access this little bit of CLA's sound.

Features

Black Lion have now managed to pack the same electronics into a double-width 500-series module, and they seem to have done so without compromise. Everything is more crowded on the control panel, of course, but they've done a great job of keeping things intuitive. To introduce gain reduction, you turn up the input, and juggle the output knob accordingly. Two more knobs adjust the attack and release times. On a typical 1176, faster speeds are achieved by turning the attack knob clockwise but on this device it operates in the opposite direction. That's the case on many other compressors, of course, and I can see the logic (turn further right for longer times), but as a regular user of an 1176 I found this mildly frustrating. The 20:1, 12:1, 8:1 and 4:1 ratio options are set using buttons, though there's no provision for the classic 'all buttons in' trick. There's also a wet/dry parallel-compression control, something that's not present on CLA's original unit but is a welcome feature in any hardware compressor now that so many of us aren't using analogue consoles. The backlit meter can be set to display output level or gain reduction. While this is a mono device, two modules can be linked for stereo operation using an RCA connector on the front panel. Finally, there's a bypass button.



True Blue

So, what does a 'blue-stripe' 1176 compressor sound like, exactly? In relation to the subsequent revisions UREI and Universal Audio released, I think of it as having more attitude — it sounds more mid-forward, and introduces a distinct tonal change that complements any compression that's being applied. This is precisely how the Bluey 500 behaved in my studio, and on vocals in particular it had a knack of presenting quite aggressive-sounding compression in a way that didn't sound overly 'messed around with'. It's a pretty addictive quality — I can understand why they wanted to recreate CLA's compressor! — and when tracking a variety of male and female vocals I found myself happily allowing the gain reduction meter to dance around with abandon.

The character of the gain reduction itself will be very familiar to users of any 1176, but coupled with this subtle mid-heavy tone it's especially pleasing. While it's famous for use on vocals, it has many other applications. I got great results hitting the Bluey 500 hard when tracking bass guitar, and it was great on snare drums and anything that needed to be brought forward in that 2-5 kHz 'intelligibility' range, as well as assertive dynamic range control.

Even if I ignore completely the enticing story of how the Bluey 500 came into being, I have to say that this is one of the most satisfying 500-series compressors I've used. 1176-style compressors are a classic for a reason, and I found this particular model to be an excellent all-round tracking compressor, and one that I would go to the bother of patching into the DAW when mixing, to help shape a vocal. Once I'd got my bearings with the controls, I didn't feel there was any compromise whatsoever from using this more compact version of the Bluey over the 19-inch one. Obviously, you need more than gear to make your mixes sound like CLA, but this is an excellent sounding compressor that's also competitively priced.

summary

Modelled on Chris Lord-Alge's favourite 'blue-stripe' 1176, Black Lion Audio have successfully ported their rackmount Bluey FET compressor into the 500-series format.

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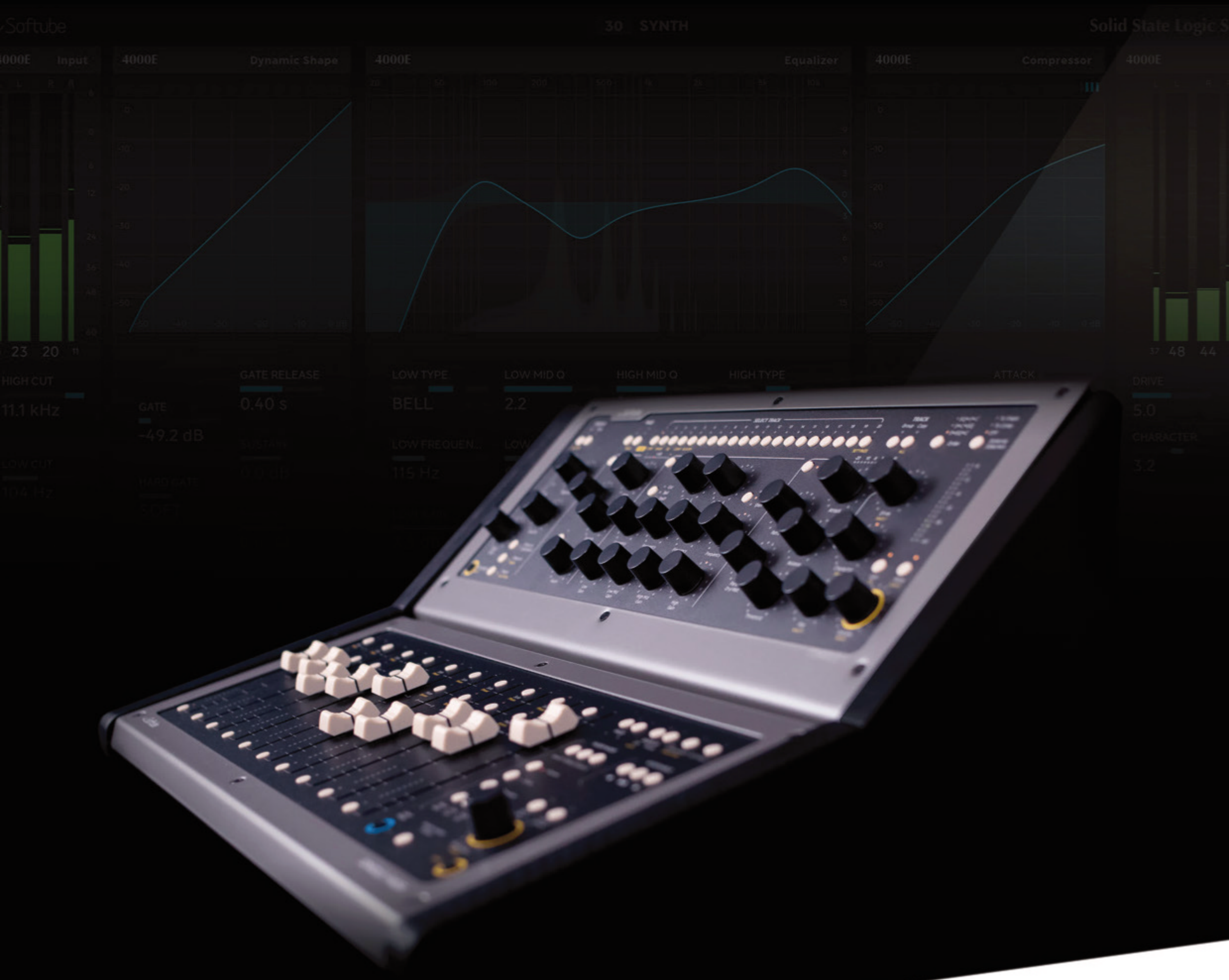
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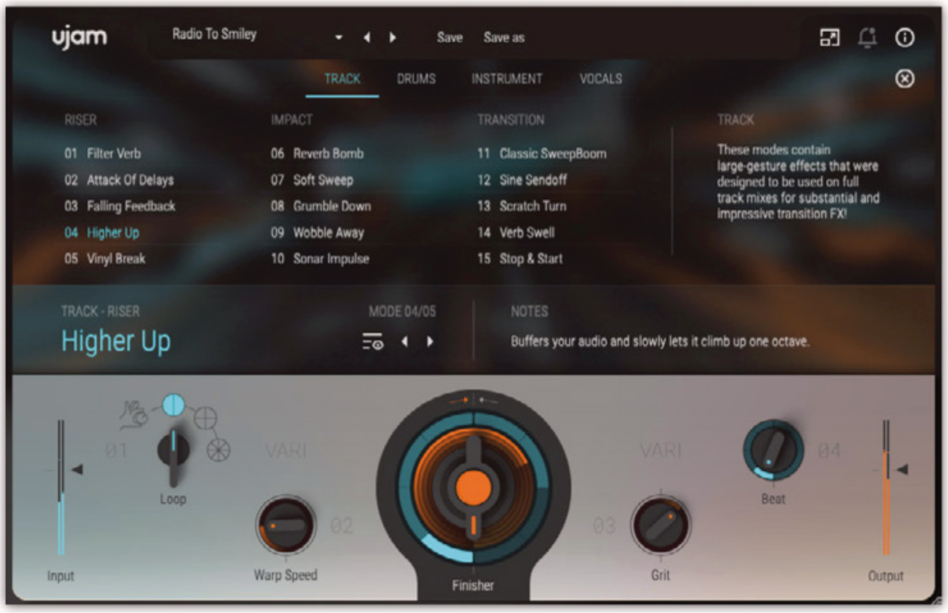
UJAM Finisher Boost



These modes contain large-gesture effects that were designed to be used on full track mixes for substantial and impressive transition FX!

Buffers your audio and slowly lets it climb up one octave.

Multi-effects Plug-in

The latest in UJAM's Finisher series aims to give your risers, impacts and transitions a boost.

JOHN WALDEN

UJAM have a design ethos that spans their whole product line: to give the user high-quality sound with a minimum of fuss. This is certainly true of Boost, the most recent addition to their Finisher line of creative multi-effects plug-ins. I have reviewed both Retro and Dynamo (*SOS* August 2021 and April 2022, respectively) and you can get a refresher on the overall concept from those reviews but, in summary, each Finisher provides different styles of effects aimed at a specific application. Each Finisher plug-in, therefore, supplies a customised collection of effects-chain presets, which can be tweaked using a compact set of macro-style controls. The concept may not appeal to the control freaks amongst us, but if you just want great results without interrupting your musical workflow, it's a winning combination.

So Macro

Boost's application is the creation of tempo-sync'ed risers, impacts and transitions (the last is essentially a riser and an impact combined) of the sonic ear-candy kind that's integral to modern music production. Once you've chosen an effects-chain preset, it can be applied to any source in your project to create a riser, impact or transition from that audio. The engine rather cleverly handles all the tempo-sync'ing automatically, so while you can control the length of the effect, you never have to worry about the precise timing. Amongst other things, the large Finisher knob controls the blend between the unprocessed and processed signals, so offers an easy way to control the intensity of the effect. And probably because the riser/impact/transition is derived from audio that's already part of your project, rather than a separate imported riser or impact sample, the effects seem to sit in an overall mix in a very natural fashion.

There are 60 effects chains in total, divided into four categories: Track, Drums, Instrument and Vocals. These labels already suggest the most likely targets for each effect, but they'll often do interesting things to other sources and UJAM also encourage you to experiment. You also get an impressive array of global presets that combine one of these effects chains with pre-configured settings for the various macro controls. These are also organised into useful categories, including risers/upsweeps, impacts/downsweeps and transitions, as well as some more off-the-wall special effects.

Each of the effects chains (and the associated presets) can be fine-tuned via the four macro-style Vari controls. Controls 1 and 4 let you adjust the length and timing of the effects, and Vari 1 also offers a fully manual mode for the effect length, for those that want it. Vari 3 and 4 control different elements that are specific to each effects chain. Labels such as Sweep, Stutter, Filter and Noise hint at what's going on, but others such as Absurdity, Shock and Vacuum just require you to experiment. Either way, despite the minimal control set, there's plenty of sonic scope here.

Ruling The Boost?

In use, Boost is fast, fun and very effective. I've tried lots of riser, impact and transition tools over the years, and in terms of getting the effects to hit their bar/beat target this is undoubtedly the slickest workflow I've encountered yet. Applied to the obvious sources (drums, synths, guitars, vocals or a full mix), the effects themselves can be as subtle or as OTT as you prefer. You can also automate many of the parameters and, of course, toggle the plug-in's bypass state, so a single instance of Boost on a track can easily be made to produce very different effects, and it's super easy to control exactly where it is used. For sprinkling transition ear candy into your electronic music productions, Boost is as easy as it gets too. There's a free trial of Boost, available on UJAM's website, and as long as you're not a complete control freak when it comes to customising your multi-effects settings, it has to be well worth taking for a spin. ■■■

summary

An excellent addition to UJAM's Finisher line, Boost makes adding tempo-sync'ed riser, impact and transition effects incredibly easy.

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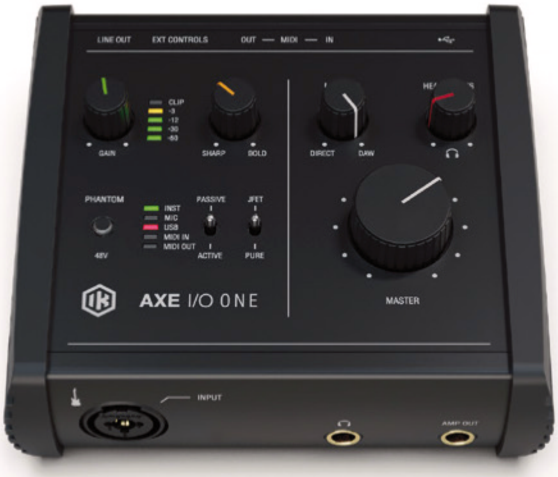
IK Multimedia Axe I/O One



USB Audio Interface

IK's mini Axe has everything you need for recording solo guitar.

PAUL WHITE

The Axe I/O One bus-powered interface is the little brother to IK's Axe I/O and Axe I/O Solo. It offers just a single combi mic/instrument input connector and three outputs, but still retains all of its big brother's optimisations for use with guitars, such as a fully discrete input circuit with switchable Pure and JFET input stages to give the choice of transparency or a little added warmth.

Conversion is 24-bit at up to 96kHz and the tech spec is impressive, with a frequency response from 11Hz to 45kHz within 0.5dB at the 96kHz sample rate option. There's also a generous collection of software bundled with the interface, most impressively IK's Tonex SE and AmpliTube 5 SE.

Physical

To keep the unit compact, the rear, top and front edges have been utilised for controls and connectors. The front edge houses the combi input, a quarter-inch TRS headphone output and the Amp Out jack. The rear panel hosts a USB-C connector, which also provides the power for the device, a security chain slot, 5-pin MIDI I/O and the two audio outputs. Two further TRS jacks can accommodate switches or expression pedals.

On the top panel, rotary controls address the input gain (with a five-section level meter alongside), Z-Tone adjustment from Sharp to Bold, monitor direct/DAW balance, phones level and master level. A button enables phantom power, with status LEDs relating to instrument, mic, USB, MIDI in and MIDI out. The last three light up automatically when a USB or MIDI connection is detected. A toggle switch optimises the input for passive or active pickups, with a second toggle switch selecting between Pure or JFET for the input when Passive is selected.

Tone Quest

IK's Z-Tone impedance-adapting input circuit is designed to present the guitar (or bass) with a load similar to that of the input stage of a real tube amplifier, which is important in retaining the correct playing feel. Here it can be adjusted to taste over a range from 1MΩ right down to 2.2kΩ. The instrument input is the jack socket in the centre of the combi connector, while the XLR section functions as a mic input with 48V phantom power, switchable either from the front panel or from the free controller app.

The dedicated Amp Out of the larger model is retained, making reamping easier, or for when using the included Tonex SE software, with which you can capture Tone Models from your own choice of amplifiers with your preferred settings and miking arrangements. There's also a generous library of Tone Modelled amps you can use right away. We ran a full review of the Tonex software in the January 2023 issue of *SOS* so I won't go into too much depth here, other than to say that it is impressive.

The Amp Out features a floating output with ground lifting switchable via software. This is independent from the two main outputs and is designed to allow the interface to feed guitar amps or stomp pedals without the risk of ground loops causing hum. As it is independent of the main outputs, it can be used to feed a DI-recorded guitar to an amplifier or stompboxes for reamping, while monitoring via the main outputs. It also simplifies using the Tonex amp capture software. Both of the main outputs are electronically balanced and may be used with balanced or unbalanced connections with no change in signal level.

In Use

Hooking up the Axe I/O One couldn't be simpler: it just needs a USB port with power, and as the device is class compliant, it springs into life without the need for drivers. However, the Mac OS/Windows control program can access all the functions of the Axe I/O One. For example, there's a switch on the control software screen that sends the instrument input directly to the Amp Out, with no latency, via a level fader and mute switch. Ground lift is also activated from here.

While having only a single input may be limiting for those who want to play the guitar and sing at the same time, I suspect that many electric guitar players will keep the two tasks separate, in which case the Axe I/O One is a perfect fit as its guitar-friendly features make it easy to dial in the right kind of sound by selecting from the Pure/JFET options and then adjusting the Z-Tone control until things feel right. As long as you ensure your input is not clipping on the loudest parts, you can then open Tonex SE and start browsing its library of amps and cabinets right away before trying your hand at capturing your own favourite settings. Given the quality of the included software, this interface has to be considered a real bargain.

summary

For recording single sources, especially guitar, the hardware and software on offer here represent very good value for money.

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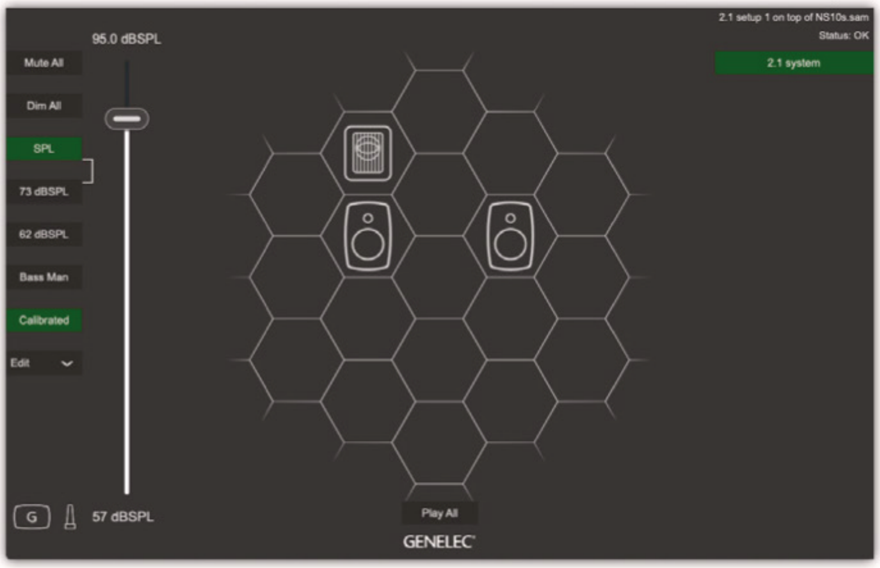

Genelec GLM 4.2

Speaker Management Software

DSP-equipped loudspeakers are only as good as the software that configures the DSP — in Genelec's case, their GLM package.

SAM INGLIS

Digital signal processing is becoming an increasingly accepted part of studio monitoring. Although it's not a get-out-of-jail-free card for poor acoustics, it can optimise speaker response for a given space in a way that simply isn't possible otherwise. This is particularly vital in immersive monitoring, where you typically have 12 or more loudspeakers that can't all be in acoustically perfect positions.

DSP optimisation can be implemented in several ways. In a software-only product like Sonarworks' SoundID Reference or Dirac Live For Studio, corrective equalisation and phase alignment are handled either by a plug-in running on your DAW's monitor output, or a 'systemwide' app that sits between DAW and audio interface. Hardware systems such as the Trinnov MC Pro and Nova process the audio between your audio interface and your loudspeakers, while some audio interfaces have calibration features built in. Or, finally, you can choose loudspeakers with built-in DSP capability.

All of these options have their own pros and cons. A software-only solution is the cheapest option, and potentially supports an almost unlimited number of EQ bands, but it can introduce practical hurdles into the music-making process. Hardware systems, by contrast, are 'set and forget', and often include hands-on monitor control; but they can get expensive, and can be complex to install.

On paper, integrating DSP into the monitors themselves offers most of the advantages of a dedicated hardware system, without the additional wiring and setup hassle. But, of course, it's only an option if your loudspeaker manufacturer provides the appropriate tools. Some monitors have no equalisation facilities at all, while others offer only basic analogue filtering to compensate for generic issues such as corner placement. And at the other end of the scale, there are speakers that have very sophisticated, network-controlled DSP engines built in.

G Force

A good proportion of the studio monitors on the market today have enough DSP built in to implement detailed corrective EQ. However, the accessibility of that DSP varies. In some cases, you'll need to calculate and transfer individual EQ curves yourself using a utility such as Room EQ Wizard or FuzzMeasure. Other manufacturers allow Sonarworks profiles to be uploaded directly to their loudspeakers, while others still have developed their own software speaker management and control utilities.

Foremost among the latter are Genelec. Most of the studio monitors in the Finnish manufacturer's range now bear the SAM (Smart Active Monitor) moniker, meaning that they have built-in DSP that can be configured over Ethernet. This is handled using a software package called Genelec Loudspeaker Manager, or GLM for short. What was initially a no-frills utility with few concessions to user-friendliness or visual design has evolved over the years, and GLM is now a pretty sophisticated package.

Although it was labelled as a mere point upgrade, the recent 4.2 release incorporated a huge amount of new functionality. In conjunction with SAM-enabled monitors, GLM will now handle speaker/room calibration in stereo and surround, bass management, some monitor control functions, loudness calibration and more.

Pop Groups

The GLM software is available for Mac OS and Windows and is free, but you'll need Genelec's Loudspeaker Manager User Kit in order to carry out measurements and calibration. This comprises a microphone and a small black 'network adaptor' that connects to your computer over USB and to your monitors over Ethernet. Helpfully, all SAM monitors come with a lengthy Ethernet cable, and you're free to connect the speakers in any order that's convenient. In order to put GLM through its paces, Genelec loaned me a 7350APM subwoofer to try with my existing 8330A monitors.

GLM detects and identifies your speakers automatically, and prompts you to update their firmware if needs be. You then define a speaker Layout by dragging each speaker to a hexagonal cell within a grid — a surprisingly intuitive and straightforward approach. Optionally, you can also enter extra information such as the dimensions of your control room. Genelec provide excellent documentation that holds your hand through the process, and includes numerous example setups.

Once you've established a Layout, you can then define one or more Groups of speakers within it. For example, in an immersive setup you might have one Group comprising all the speakers, another for 5.1, a third for 2.1 and so on. Each Group can have its own calibration settings. Again, the

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» process is very user-friendly, and guided at every step by helpful prompts.

Two important decisions have to be made at the start of the calibration process. You can choose whether the calibration should be based on a single measurement at the listening position, or whether it should be a 'multi-point' calibration that takes into account additional measurements at different positions. You can also select whether each speaker should have its own individual EQ, or whether symmetrical left-right pairs should be equalised identically. In an ideal control room, the latter is preferable, but the Individual option can be helpful in irregular rooms or where the listening position is not central on the front wall.

You can then begin the AutoCal measurement process, which involves the usual sine sweeps played through each speaker in turn, captured on the measurement mic at or around the listening position. When you've taken a sufficient number of measurements, the data is uploaded to the Genelec Cloud to be crunched into suitable EQ and delay values. If you're using a subwoofer, you can also carry out an optional AutoPhase calculation, which plays a series of chirps through the sub and one of your satellites in order to optimise phase alignment. Finally, you can then store the resulting settings within the speakers themselves. Should you wish to inspect or manually edit the EQ settings for each speaker, clicking on it will bring up a pop-up window where you can do exactly that.

Go GLM?

As I said at the start of this review, an integrated system from a single manufacturer has the potential to offer both ergonomic and functional advantages over hybrid systems involving plug-ins or additional hardware. GLM 4.2 bears this out, and if it was a separate commercial product in its own right, Genelec would easily be able to justify charging top dollar for it. As it is, you have to pay for the User Kit to get the benefits, but that compares pretty favourably with existing alternatives. For example, Sonarworks' SoundID with a measurement mic costs €299 for the stereo-only version, with the multi-channel edition coming in at €549, while the stereo-only IK ARC3 is €249.99 with measurement mic. The MA1 calibration kit for Neumann DSP monitors is £239, but is currently stereo-only. GLM supports all surround configurations at no extra cost, controls many other functions such as bass management, and eliminates the hassle of using plug-ins to perform speaker equalisation — though, admittedly, you don't get SoundID's headphone correction capabilities or ARC3's virtual monitoring.

I can't say how the GLM calibration compares with third-party options sonically, but in my space, it made a subtle but wholly positive difference compared with using the speakers in their uncalibrated state. And although the 8330As are nominally flat down to 40Hz, the difference that adding the subwoofer made was palpable, even at low volumes. They say that you shouldn't be aware of the existence of a sub until it's switched off, and GLM's calibration and bass management achieved that effortlessly. The 8330As are fine speakers anyway, but GLM makes them even better.

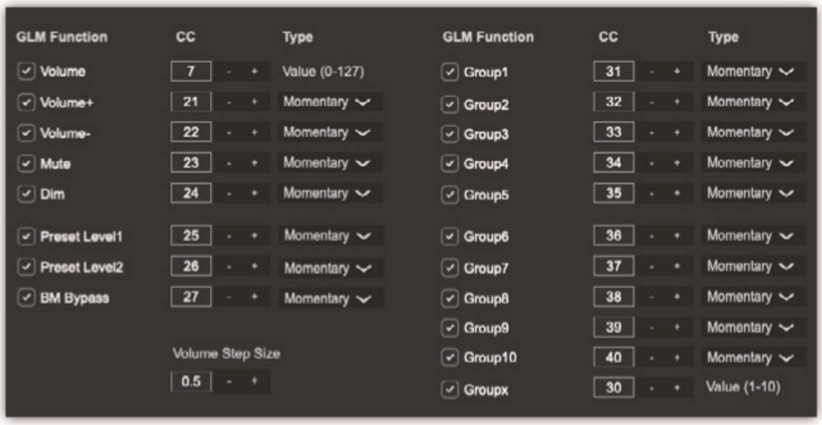
GRADE School

One of the innovations introduced in GLM 4.2 is the Genelec Room Acoustic Data Evaluation or GRADE Report. As long as you're logged into your Genelec account while you carry out the measurement process, you can opt to receive a PDF document that summarises the performance of the monitors within your room. This goes into much greater depth than I expected, and is a genuinely valuable resource. Not only does it dive into the frequency response of each monitor in considerable detail, but you also get thorough evaluations of time-domain performance and reverberation characteristics. There's a slew of easily comprehended graphs including waterfall, wavelet, decay time and early reflection plots, and any potential problem areas are clearly highlighted. If you do feel that the intervention of a professional acoustician is desirable to sort any of these problems out, the GRADE Report would be an excellent document to share with him or her as a starting point.

Monitor Control

As GLM has developed, Genelec have added features such as the ability to calibrate monitor Groups to a reference SPL, to store volume presets, adjust the level in user-defined increments and enable or disable bass management. GLM 4.2 brought a further improvement in the shape of MIDI support. It's now possible to assign MIDI Continuous Controllers to all of these parameters, so you could for instance use a MIDI control surface to adjust volume, switch between monitor Groups, mute or dim the speakers and more; and Genelec offer a simple hardware controller of their own for level adjustments. For some reason Volume defaults to MIDI CC20 rather than 7 as you might expect, but all of these assignments can freely be changed by the user. A MIDI Learn function might be a nice addition in a future update.

GLM doesn't sit in the audio path, so can't handle things like mono fold-down, talkback, secondary speaker switching or headphones. But if you want simple level adjustment with mute/dim and the ability to switch easily between stereo and surround, it does so with minimal fuss. I didn't have Genelec's own volume control to try, but anything that can generate CC data seems to do the job.



GLM Function	CC	Type	GLM Function	CC	Type
☑ Volume	7	Value (0-127)	☑ Group1	31	Momentary
☑ Volume+	21	Momentary	☑ Group2	32	Momentary
☑ Volume-	22	Momentary	☑ Group3	33	Momentary
☑ Mute	23	Momentary	☑ Group4	34	Momentary
☑ Dim	24	Momentary	☑ Group5	35	Momentary
☑ Preset Level1	25	Momentary	☑ Group6	36	Momentary
☑ Preset Level2	26	Momentary	☑ Group7	37	Momentary
☑ BM Bypass	27	Momentary	☑ Group8	38	Momentary
			☑ Group9	39	Momentary
	Volume Step Size		☑ Group10	40	Momentary
	0.5		☑ Groupx	30	Value (1-10)

Monitor control features in GLM 4.2 are controllable over MIDI.

summary

GLM has quietly evolved into a seriously powerful speaker management system with very sophisticated calibration and room analysis tools. It's superbly documented and very easy to use.

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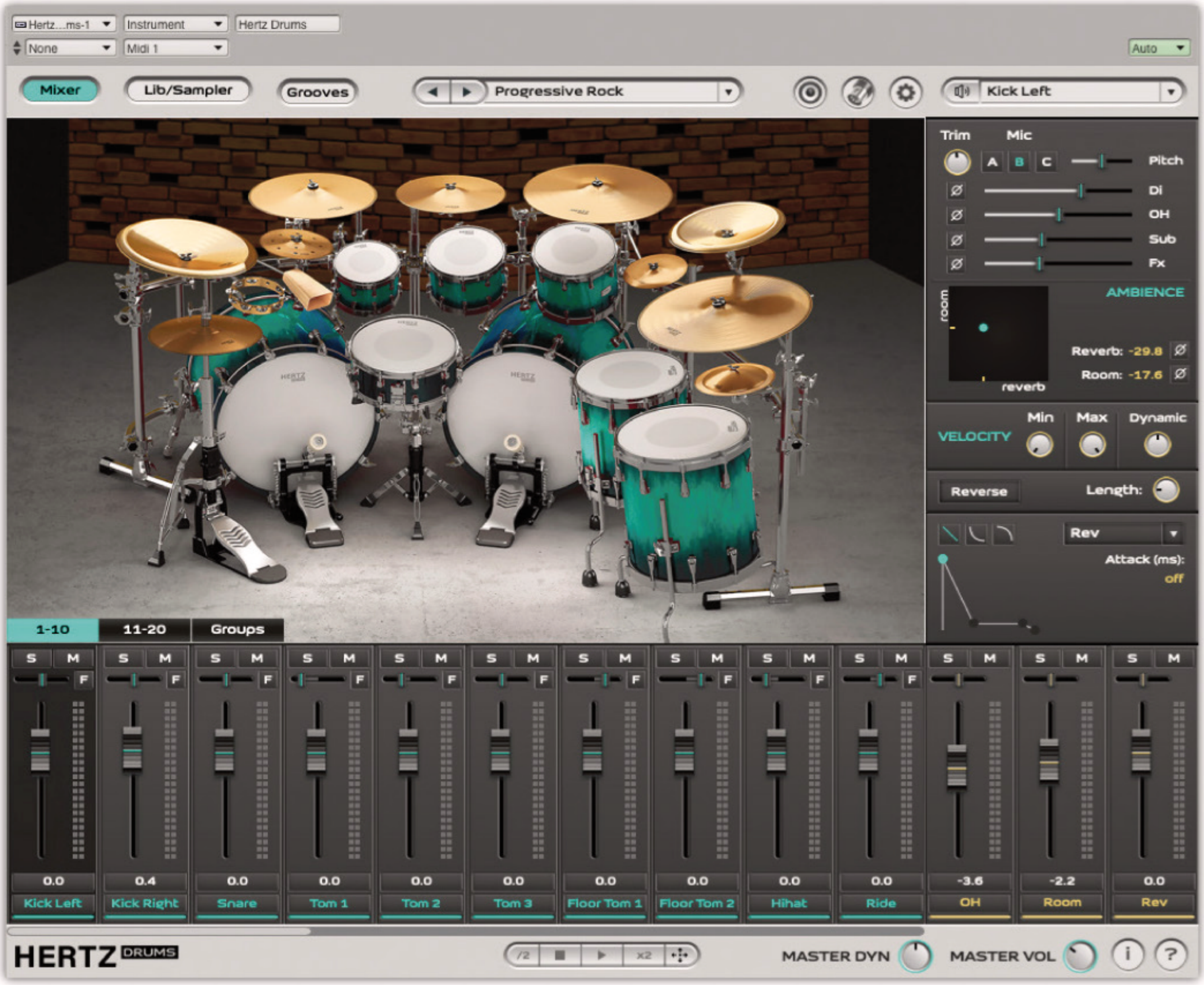
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Trim
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A
B
C
Pitch
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OH
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Fx
AMBIENCE
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reverb
Reverb: -29.8
Room: -17.6
VELOCITY
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-2.2
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Kick Left
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Sample Libraries

If you like your drums heavy, the Hertz Drums Bundle might be just the thing.

MARK GORDON

Virtual instruments have become a part of everyday life for computer-based music producers, covering many different instruments, styles and genres. Much like the iconic phrase from Apple back in 2009, “there’s an app for that,” nowadays “there’s an instrument for that.” Drums being a fundamental building block of modern music production and most people not having the facility (or budget) to record them, there are lots of virtual drum instruments on the market — often including a virtual drummer.

Hertz Drums are new kids on the block, and not a company I was previously aware of in what is a crowded market. Their instrument is the brainchild of brothers Wojtek and Slawek Wieslawski, owners and in-house producers at Hertz Studios in Poland. Renowned for their work with many top metal bands such as Behemoth, Vader and Hour Of Penance, the brothers combined their knowledge and experience with that of software developer Alex Yashchaka, and Hertz Drums was born. It’s pitched unashamedly at a rock and metal audience, but Hertz Drums offers a great deal more than blast-beat kick drums and gunshot snares, as we will find out.

Mega Hertz

The Hertz Drums plug-in comes in a familiar ‘instrument plus expansion packs’ format; you purchase the plug-in instrument host, then add the expansion packs that best suit your needs. It’s worth pointing out that the basic instrument doesn’t ship with any sounds, which is perhaps a little unusual, so you’ll need to buy at least one expansion pack to get started. My review copy is the Hertz Drums Bundle, which includes the plug-in instrument plus three expansion packs, slightly uninspiringly named Red, Blue and White. These cover “classic rock,

Hertz Drums Bundle

£199

PROS

- Great-sounding drums ‘out of the box.’
- Extensive editing options.
- Three mics per kit element.
- User-friendly graphical interface.

CONS

- Potentially limited genre appeal.
- Included grooves a little uninspiring.
- No song creation or MIDI groove editing facilities.

SUMMARY

If you’re producing heavier guitar-based rock and pop, Hertz Drums offers a selection of well-produced, great-sounding drums that are a perfect fit for this type of music.

funk, alternative, progressive, classic metal and many more”. I found installation trouble-free, and the process of pointing the plug-in to where the expansion packs are installed is equally simple.

On opening the instrument, you’re presented with the main screen, which depicts an imposing double-bass-drum kit surrounded by five toms and a multitude of cymbals, along with a tambourine and the ubiquitous cowbell. This very much sets the scene for where these kits sit in the musical landscape, and it isn’t country & western!

Beneath the kit is the mixer, and to the right are editing tools for shaping the drum sounds. The mixer comprises 20 individual faders (one for each kit piece) and seven group faders covering kick, snare, toms, hi-hat, ride, cymbals and percussion/misc levels. Three further faders provide control of the overhead, room and reverb levels. Each fader is equipped with mute and solo buttons, plus a Follow button that ensures room, reverb and overhead panning is kept ‘in line’ with any adjustments made to the direct signal.

Across the top of the window are tabs that select the Library and the comprehensive Sampler sections of the plug-in (see the ‘Taking A Sample’ box), and also the MIDI Grooves player window. When selected, these elements appear in place of the mixer below the kit. Sample packs and preset kits are selected from a drop-down menu, with typical load time a very passable 20-30 seconds on my old 2014 MacBook Pro. It’s nice to have everything available in a single window and this design makes for a very usable interface, although unfortunately it is a fixed size and isn’t currently scalable.

Drums, once loaded, can be auditioned by clicking on them with the mouse or via MIDI input from a keyboard, electronic drum kit or other MIDI input source. Hertz Drums has a very neat trick up its sleeve when it comes to the tricky issue of which notes are mapped to which drums, and includes several presets that replicate the MIDI mapping used by popular electronic drum kits and software instruments such as Superior Drummer, Steven Slate Drums 5, Addictive Drums and Logic Pro X. In practice, this means that if you’re used to a different drum plug-in you don’t have to remap the drums or learn a completely new set of keys. You can, of course, create your own MIDI maps, but Hertz cover many of the most popular options

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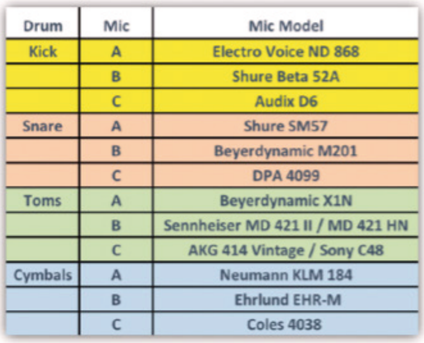


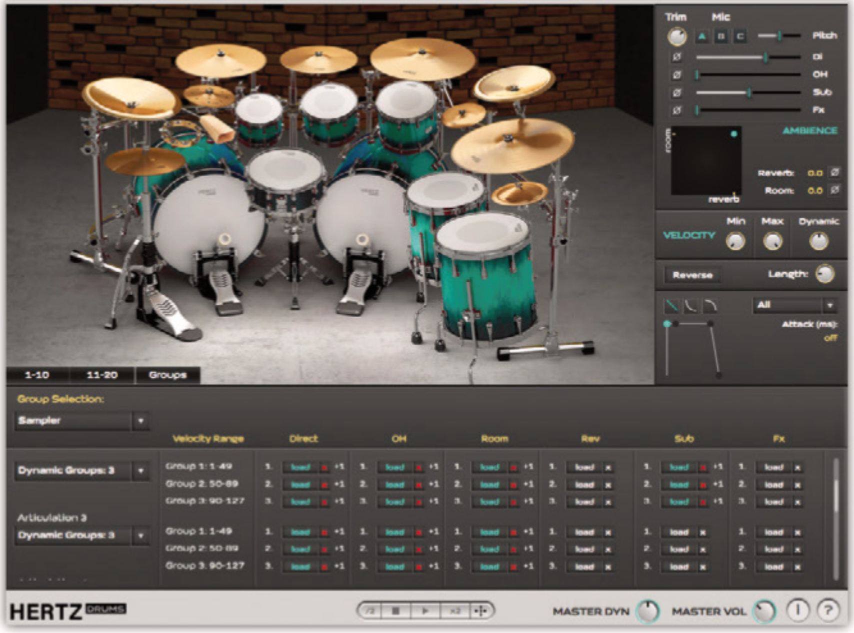
Drum	Mic	Mic Model
Kick	A	Electro Voice ND 868
	B	Shure Beta 52A
	C	Audix D6
Snare	A	Shure SM57
	B	Beyerdynamic M201
	C	DPA 4099
Toms	A	Beyerdynamic X1N
	B	Sennheiser MD 421 II / MD 421 HN
	C	AKG 414 Vintage / Sony C48
Cymbals	A	Neumann KLM 184
	B	Ehrlund EHR-M
	C	Coles 4038

Three different mics (A, B and C) were used to record each of the kit elements, all routed through the Hertz Studio Neve console.

Taking A Sample

Many drum virtual instruments give you the option to import your own samples to replace an existing drum sound or to layer with an existing kit element. Hertz Drums takes this to another level by enabling any of the drums to be assigned to a user sample that can have six dynamic layers and eight articulations. You can also import individual samples for the direct signal, overheads, room, reverb, sub and FX — meaning that you can configure your own kit with the same level of detail as the included kits, and edit it in the same way, using the ADSR and pitch controls.

With so many options, in terms of velocity layers, articulations and sources, it’s an intensive process to create a whole kit from scratch. I managed to make a passable ‘John Bonham’ kit using a selection of samples from Superior Drummer that included room and overhead elements, but if you’ve purchased a plug-in that comes with such well-produced libraries as Hertz Drums, it would seem a little pointless to spend hours creating your own. I suspect what these user sample powers will be best suited to in practice is allowing you to grab a drum or cymbal from another sample pack to replace or augment an existing kit element, with all the flexibility and editable parameters of the original.



When creating your own kits using imported samples, you can include an impressive six dynamic layers and eight articulations per drum. The same number of samples can also be imported for the overheads, room, reverb, sub and FX slots.

» and may well add other MIDI maps in future updates.

No Silence In The Library

The Red, Blue and White libraries all weigh in at around 5.5GB, and each features five snares, five kicks, five toms, one set of hi-hats, three china cymbals, three splash cymbals and three crash cymbals. These are configured into preset ‘genre’ kits in each library. Examples include Classic Metal, Power Metal, Art Rock, BritPop and Stoner Metal, each kit being comprised of a drum and cymbal combination that suits the genre in question.

Kit elements are made up of several different samples, or ‘articulations’, which include centre, rimshot and shallow hits from both the left and right hands, and each of these articulations features multiple dynamic layers. (Cymbal articulations include bow, tip, bell, edge and choke dynamic layers). Each articulation is assigned to its own MIDI note so, with careful programming, drum parts can sound very natural and realistic. Adding further sophistication, all the drums and cymbals were recorded through Hertz’s Neve console using three different mics (A, B and C), giving access to a fantastic range of subtle sound variations.

Within the Lib/Sampler window each available drum sample is represented by a white square organised into three rows that represent the A, B and C mic variations. As you click on each square to audition the drum, its description (Birch 12-inch Tom, Custom Dark 18-inch Crash, and so on) appears at the top of the window. When you’ve identified the element you like, you can simply press the Load Instrument button to place it in the current kit. Initially this looks like quite an odd way to view and audition a drum library, but in practice I found it worked incredibly well — a great way to quickly audition the various kit pieces and hear the subtle differences created by the three mics.

Total Control

To the right of the main kit screen are the instrument edit parameters. Clicking on any kit element loads its settings and displays its name at the top of the window so you know exactly which drum you’re editing.

The trim knob and pitch sliders (as you might expect) adjust the level and pitch of each drum, and the A, B and C buttons switch between the three mic options. Next are four sliders that adjust the DI, OH (overhead), Sub and FX level of each drum. DI controls the level of the direct mic and the OH slider adjusts how much of the selected drum is present in the overhead mics routed to the overhead group fader in the mixer.

In the case of toms and kick, the Sub slider controls how much signal from an additional low-frequency sub mic employed in the recording process is introduced, while if a snare drum is selected the same slider controls the level of the bottom mic on the drum. The FX slider introduces an additional sample to the overall drum sound, which can be blended in to help create the unique Hertz drum sound found on many recordings made in the legendary studio. All four faders have individual polarity invert buttons that could be very useful in overcoming any problems when importing and combining your own samples with existing Hertz drums.

The Ambience section of the window uses an X/Y grid to control the amount of ‘room’ and reverb applied to each drum, with the room element being the actual sound captured in the Hertz studio during recording. As with the overhead level, room and reverb signals are routed to the corresponding faders in the mixer. Polarity invert buttons are also provided for each of these parameters.

The Velocity knob allows you to dial in a minimum and maximum level for each drum, to control dynamic range. Although this is actually dictating which sample layers are being used, its function is similar to applying a compressor to an analogue signal.

Finally, there’s the ADSR curve feature, whose intuitive interface enables you to shape the attack, decay, sustain and release of a drum or cymbal or, by way of a flip menu, the individual reverb, room, direct or overhead elements independently. This is a great facility that lets you tailor drum sounds in fine detail. A Reverse button is also provided, should you fancy a backwards cymbal or snare effect.

Blast Beats

Grooves have become a big part of virtual drum instruments, and Hertz drums doesn’t shirk its responsibility in this department. An impressive 1560 grooves are included, divided between intros, beats, fills and ends, and covering funk, rock and nu-metal genres. Styles, time signatures and tempos are displayed to the left of the screen, and the grooves themselves are auditioned by clicking on white squares, in the same way that kit elements are auditioned.

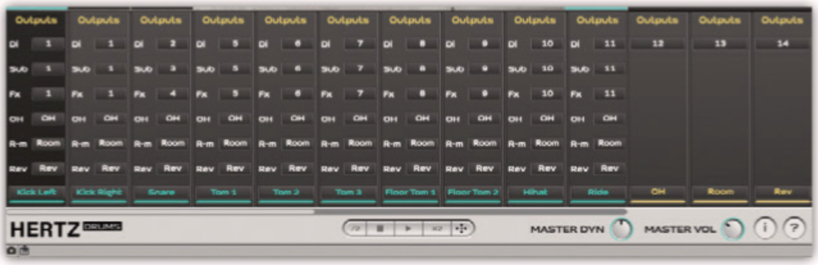
Beats are divided into Hi-hat, Ride and Mixed categories, and fills into Snare, Toms, Snare + Toms and Mixed categories, with the complexity and intensity of each increasing as you move from left to right.

Hertz Drums syncs to your DAW tempo, so the list of tempos is merely a guide to the original recorded bpm

Beat Routing

Even though the sounds within Hertz Drums are ‘song-ready’ and can be edited and processed extensively inside the plug-in, it’s often useful to be able to process each element individually with your DAW. The Audio Routing window offers a hugely flexible matrix of 16 outputs that can be assigned to not only the individual drums, cymbals, reverb and room outputs, but also to each of the direct, overhead, sub and FX signals individually.

Typically, you would route all the elements of a drum (direct, sub and FX) to a single output (which can be accomplished by holding down Alt when selecting the output) and send the overhead, room and reverb signals to their own channels in your DAW. Hertz Drums allows you to send the individual direct, sub and FX signals to their own outputs to be processed independently within your DAW. I can’t see this being something you would often want to do but having the option available is potentially useful.



	Kick Left	Kick Right	Snare	Tom 1	Tom 2	Tom 3	Floor Tom 1	Floor Tom 2	Hihat	Ride	OH	Room	Rev
	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs	Outputs
DI	1	1	2	5	6	7	8	9	10	11	12	13	14
Sub	1	1	3	5	6	7	8	9	10	11			
Fx	1	1	4	5	6	7	8	9	10	11			
OH	OH	OH	OH	OH	OH	OH	OH	OH	OH	OH			
R-m	Room	Room	Room	Room	Room	Room	Room	Room	Room	Room			
Rev	Rev	Rev	Rev	Rev	Rev	Rev	Rev	Rev	Rev	Rev			

Comprehensive routing allows for the direct, sub, FX, overhead, room and reverb signals of each kit element to be individually routed to one of 16 individual outputs.

of the grooves. Any groove can be auditioned at half or double time by selecting the corresponding button in the transport. Unlike Toontracks' EZ drummer, Hertz Drums doesn't have the facility to create songs or edit grooves, so you will need to drag your selected beat to a MIDI track in your DAW.

The grooves are well played and there's no hiding the genre in which this instrument is positioned — no hip-hop or half-time shuffles here! Initially, it can feel as though there's a lack of subtlety in some of the grooves, and they definitely suit the heavier end of the drumming spectrum. However, once you drag them into your DAW and look at the MIDI data, you can see that the many dynamic levels and articulation have been put to excellent use.

In addition to the installed grooves, Hertz Drums also offer a 'Not 4/4' MIDI pack that features (you've guessed it) 520 grooves in time signatures ranging from 5/4 to 19/16 — should that be your thing!

Love Hertz?

Full disclosure here — I'm a child of the '80s, and metal in its various forms isn't really my musical genre of choice, but I do like great-sounding drums, and Hertz Drums certainly offers plenty of those!

Right out of the (virtual) box, the drums sound excellent. Overall, they have a processed quality, but in a good way, and certainly one that lends itself to the rock and metal genres that they are primarily intended for. You won't need to spend hours EQ'ing and compressing these drums to achieve professional results that cut extremely well through a mix of heavy guitars and basses. The Alt Rock and BritPop packs in the Blue library have a slightly less aggressive character and would sit well (as the names suggest) in guitar-oriented pop, making this pack perhaps the best option for general use.

I was impressed by the level of detail available in the edit windows. The ability to audition three different mics is not something I've come across before, and the subtle yet noticeable differences they lend, in conjunction with the other editable parameters, significantly augment your sound palette.

As for the included MIDI grooves, there are certainly plenty to choose from and I was pleasantly surprised by how much I enjoyed the method of auditioning them. As a drummer, I tend not to use pre-programmed beats and find that they invariably don't fit perfectly with the song you are working on, but they are certainly great to jam to or as a basis for song creation.

The sampling is a neat and feature-packed addition to the plug-in that I doubt many people will take full advantage of, but it's an impressively detailed way to add your own stamp to a kit. Interestingly, Hertz Drums also ships as a 'Lite' version, which keeps all the features of the Standard package and only drops the sampler — possibly the better option for someone simply wanting to add this type of drum sound to their armoury.

Very clearly aimed at the metal market, Hertz Drums is a great-sounding and well-thought-out plug-in that offers an exceptional level of detail and editability. Metal music can be a pretty serious affair, but this plug-in is a lot of fun!

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TalkBack

Ross Dorkin

WILLIAM STOKES

"The joke, for me, is that on some level, you only become a producer when you start calling yourself a producer," laughs South African musician and producer Ross Dorkin. "I probably realised a bit too late, maybe six to eight years into doing it, that I was actually producing!" Having grown up in the coastal town of Pietermaritzburg, it was while studying in Cape Town that Dorkin met future bandmate Matt Field. Together with drummer Robin Brink they would form the band Beatenberg, fusing indie rock and pop with distinctive South African rhythms and playing techniques. Arguably the band's biggest break came courtesy of Mumford & Sons' *Johannesburg* EP, a collaborative body of tracks recorded with Beatenberg, celebrated Senegalese singer Baaba Maal and Malawian trio the Very Best.

Since then Dorkin has diligently cultivated his career as a producer and engineer alongside his work as an artist. Having travelled to London to undertake a Masters degree at Goldsmiths University, specialising in spatial audio composition with Dolby Atmos, Dorkin continues to live and work in the city, but remains "very much plugged into the African continent".

At the moment I can't stop listening to

There are at least two things that I can't stop listening to. One is Oklou, she's a French alternative pop singer. For me, she's just a got completely unique voice, harmonically and timbrally. She's got a mysterious sort of background — you don't really know exactly what she studies or what she's about. But if you see her show, or hear her *Galore* recordings, they're a beautiful, unique approach to harmony and melody. I think, in the same way that Caroline Polachek and Rosalía have grown in the last three or four years, she's the next wave.

With Rosalía, it all just comes back to rhythm, you know, which is a big thing for me. And I don't want to say it's just because I come from where I come from, but those are the idiosyncrasies that you bring to different countries. More than almost anything else, it's your sense of rhythm, your sense of groove, your sense of syncopation: that feels very endemic to where we're from, and it's not something that you can really fake. I think with Rosalía; the rhythmic identity, the melodic focus, the production focus, they're never superficial. There's never anything unnecessary. It's just so beautifully executed, always.

The project I'm most proud of

I think, probably, the last Roo Panes record [*The Summer Isles*], just because I had so much fun doing it. I think we got a lot of really beautiful moments recorded on that. It's very acoustically focused. From an engineering and a production perspective, it was something where I just felt like I got the sounds that I wanted. It was the most ideal record making process, really: we literally just went in, recorded the sounds and then barely had to mix it! And it was done! I think we did most of the mixes in one afternoon, on the final day. I did one revision, and I had, maybe two notes. For the whole record. That's how smooth it was.

It was also the process, as well, just being in the studio. We just had a really good time and enjoyed it so much together. And I felt like I made really good friends with the guys I was working with. So it was a real pleasure. It's funny, you can be proud of the result of something, but you can also be proud of something because of the experience that you had. And the experience the artist had too. It's not always just about the sonic product, if that makes sense. Right from the first test sessions that Roo and I had, we just got a sense of ease with each other and I felt we could be really honest. We could trust each other's decisions without having too much conversation about what the reverb sounded like, or whatever. For example, Roo doesn't really like having any bass on the recordings. I think I played, like, three bass notes on the whole record! And It was quite fascinating to work in that way. For me it was so stimulating, in such a different way, I suppose. And I really do honestly feel like production is about making someone feel like they can be more themselves. I hope that he felt that way about this record. I think he did.

The first thing I look for in a studio

Does all the gear work? That's probably the first question I'll ask: what works and what doesn't, because there can be so much superfluous equipment around that just doesn't function. And maybe that's cool. I can dig it. You can work around that stuff. But, man, I'll just need to ask, "Is there a good basic vocal chain? Do I have a set of Coles 4038s?" I can do a lot with three mics. I just need a good vocal chain: a [*Neve*] 1073... An SM7, for that matter! It doesn't have to be a fancy vocal mic, but you need a good vocal chain and at least a clean stereo pair, and some extras that actually work. That's key, for me.

The person I would consider my mentor

I think it would be my childhood music teacher, Jimmy Freeborn, who I studied with for eight years. He was certainly a mentor for me in terms of getting into music and thinking about music in this dynamic and open, and — for lack of a better term — a spiritual way, reflecting on the spiritual dimension of music. He had a really balanced view on genre, on approaches to making music, and on why you even make music, as a person. Most of our lessons, after six years, were more conversations than they were about learning anything! He taught me guitar, bass, music theory, composition... He was quite a mad guy. He looked almost like a doppelgänger of John Lennon. He had a massive, incredibly well organised vinyl library, a cassette library, a CD library, and also an actual library of books. Everything was meticulously catalogued. So if I asked for, say, a certain kind of choral music, he would give me a mix CD of that. If I was like, "I want more of that," he'd give me five more CDs! I was 13 or 14 years old, getting into things like the Residents and their album *Duck Stab*. And just lapping it up. You know, that's the kind of thing that not many 13-year-olds in South Africa have access to. He had actually played with my father, who was a musician for 25 years, playing in bands. My father passed away when I was 10, and I only started playing music a few months after my dad died. So his best friend was my mentor.

My go-to reference track or album
It's probably gonna be 'Malamente' by Rosalía. That on the speakers always gives me a very good indication of things in the room. And also, more recently, some of the tracks from Marcus Mumford's new record that he made with Blake Mills, especially the first couple of tracks. They're just a really good indication of a good vocal sound and a good acoustic guitar sound. And then, probably Caroline Polachek. Probably like 'Bunny Is A Rider', or something like that. That also gives me a good feeling for the room I'm working in.

My top tip for a successful session
Leave your shit at the door! Don't let your stress influence the temperature of the room. If you're able to keep talking and keep the energy up without anyone knowing what's going wrong then you've won. And there might be some legitimate issues going on. It's a lesson every day. And it's hard, but that is the job. It's keeping the energy good and relaxed, no matter what happens. And things are gonna happen! I think that's the actual draining pressure on engineers or producers, on a long-term basis. You are the temperature in the room, all of the time. You have to keep up that energy, and people expect that of you. And, I think, rightfully so.

The studio session I wish I'd witnessed
Oh, well: *Voodoo* by D'Angelo. Obviously! To get more specific: 'Untitled (How Does It Feel)', that track. Can you imagine just being there! The best productions and recordings, for me, feel like they couldn't possibly have been made in any conceivable, physical, brick-and-mortar room on this planet. It just feels like there's no way you could visualise that magic, of what that room could actually be in reality. It's so beyond that. It's intimate, but it's expansive. Deep, which is a very hard thing to get right. But it's dry and close. It never feels like it's lacking, never feels like it's wanting for anything.

The producer I'd most like to work with
At this point, probably Blake Mills. He seems like the kind of person who would allow people to go wherever the hell they wanted, but still be able to get an incredibly unique, almost iconic result.

The part of music creation I enjoy the most
Is it cheesy to say it's just when you enjoy the process itself? I'm trying to find a better way of wording it. I guess what I'm trying to say is that I'm less interested in the final product, and more interested in the process. Because I know that if the process is good, the final product will be good. And I know if the artist enjoys the process and I enjoy the process, and it's a good energy in the session, then even if a vital part isn't good, it's still gonna be good. Because the whole feeling towards making the music will be good, and it will make you want to make more music. That's my feeling towards it. If you get that right, then everything else will follow naturally.

The advice I'd give myself of 10 years ago
Don't always think that everyone else in the room knows more than you. I think impostor syndrome is a difficult thing to deal with, in different environments and with different engineers, but try to find a way to maintain being yourself, asking questions when you need to, but don't always assume that everyone knows. It's about finding a balance between having a thirst for knowledge and not feeling like you're constantly the least knowledgeable person in the room. Your ideas are always valuable, regardless of your technical knowledge. ■■■

Elk Live

Remote Collaboration Service

With their clever audio networking technology, Elk Audio aim to make low-latency, long-distance collaboration a reality.

CHRIS TIMSON

It's 2020 and the first lockdown has just begun. My wife Anne and I sing in a three-part close-harmony group called Eagle Alley, with our friend Rob who lives 12 or so miles away. Our group cannot of course meet to practise, but really want to. We've experimented with Zoom and learned for the first time the awful impact of the word 'latency'. After a lot of searching we discover an open-source package called Jamulus, which has somehow managed to minimise latency sufficiently to permit us to sing together, and gratefully we resume rehearsing, but now over the Internet.

I was sufficiently impressed with Jamulus to write a feature about it for *Sound On Sound* (Setting Up & Using Jamulus, September 2020). Even though lockdowns are (we all hope) a thing of the past, Rob has recently moved rather further away, and so we're still using Jamulus from time to time for rehearsals. Consequently, I've been keeping an interested eye on low-latency audio software, and so was much intrigued when offered the chance to review the Elk Live Bridge, a hardware solution from Swedish company Elk Audio which is claimed to offer very high performance whilst remaining extremely easy to use. Elk Audio have devised their own version of Linux called Elk Audio OS, which is heavily optimised for low-latency audio and which lies at the heart of the Elk Audio Bridge.

Well, that's how this review started, but I have to tell you that halfway through, the review changed completely. Initially, I was looking at a very interesting, if expensive, hardware approach for low-latency music-making over the Internet. By the end, I was exploring a whole ecosystem of hardware and software alternatives dedicated to this end, which is temporarily free of charge in its software aspects (you still have to pay to buy the hardware). So: how to proceed? I will start with the hardware route, because that's how I met it and the user experience is remarkably similar in both approaches.

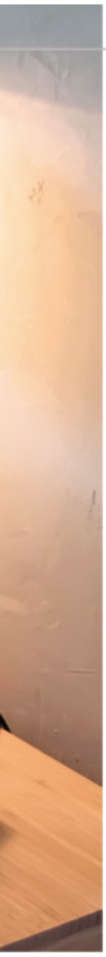
Bridging The Gap

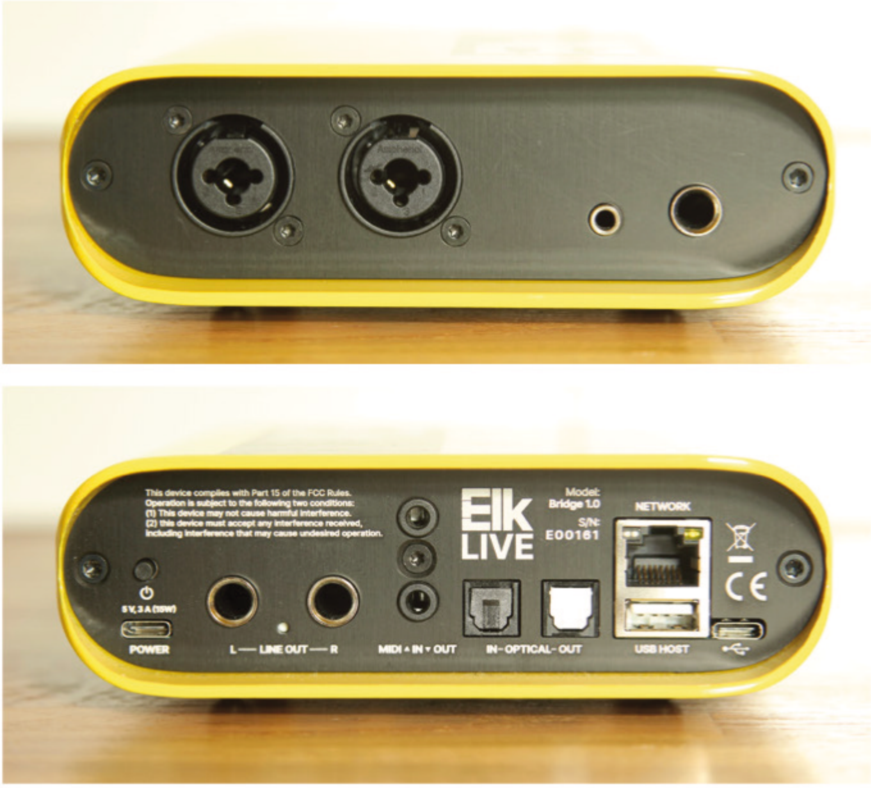
At first glance, and indeed at second or third glance, you might well mistake an Elk Live Bridge for a conventional audio interface. The solidly made and attractive metal case has two XLR/line combi inputs on the front plus two headphone sockets and, on the back (in order), USB power in, left and right line outs on quarter-inch jack sockets, two mini-jack sockets for MIDI I/O, two optical I/O sockets, an Ethernet socket, a USB host socket and finally a USB data socket. Then you might notice that apart from a small on/off button there are no physical controls at all, and no LED meters or indicators either (in fact there are some LEDs, but they're well hidden underneath).

The package contains the Elk Bridge itself, along with a natty carrying bag for the unit, a short Ethernet cable and a PSU. That's it: there's no printed documentation of any kind. However, look carefully in the box and you'll find this text printed on part of the packaging:

"No paper manual?! Nope. We hate paper cuts and want to make sure you get the latest and greatest. You can find everything else you need at www.elk.live."

So off you go to the Elk Live website, where you will find the information you need to get started, the gist of which is this. Unlike a conventional audio interface, the main connection your Elk Bridge makes is not to a computer but to your router. This connection must be by Ethernet cable, as is normal with low-latency products for the Internet — Wi-Fi has too many latency problems of its own to be usable. The data USB port is not necessary for the use and control of the Bridge (although more about what it's





The Elk Live Bridge has much in common with a typical audio interface. On the front panel are two XLR/jack combi inputs and two headphone outs, while the rear panel houses USB power and host ports, stereo line outputs, MIDI I/O on mini-jacks, optical I/O ports, and an Ethernet port for connecting to your router.

there for later). You don't need to install driver or control software on a computer either: everything is done via the Elk Live website. The other significant requirement is a decently fast Internet connection. Elk Audio recommend a minimum 10Mbps up/down speed, and ping less than 10ms. They also suggest that all participants in an Elk Live session should be within a 1000km (620 mile) radius of each other in order to achieve good latency (that's pretty good, actually).

Like & Subscribe?

Once you've plugged your Elk Bridge PSU into the mains, connected up your Ethernet cable to your router, plugged in a mic or an instrument and attached a pair of headphones, it's time to set up an Elk Live account. This is where things are starting to change. Creating an Elk account is the first step in setting up a subscription, which is essential as you can't join an Elk Live session without one. When I started this review, subscriptions cost £13 a month, so in our case, with three people, that potentially meant paying £49 each and every month to use the service. The big change is that, since the beginning of April 2023, subscriptions are free while the company ponder the best funding mechanism for the future. Much more about this below.

Note that for all web interactions, Elk Audio recommend the Chrome browser as that's what they use in development, but I used Firefox throughout this review with no problems. Also bear in mind that although the Bridge is connected to your router and not your computer, you will still likely need to have your computer near the Bridge. This is because your mics and instruments run on the Bridge but the control software is a browser running on your computer. So, having set up your Bridge, your account and your subscription it's time to go to the Elk Live web page...

Note how at the top left-hand corner of Screen 1 it says "Bridge status: All good to go!" If your Bridge is not connected or switched on it will say "Bridge not found". All being well, the first thing you will want to do is contact your band colleagues, who will also have Elk Bridges and have set up Elk Live accounts. This process is called Add Friends. For this, you supply the email address of a person with a Bridge and Elk Live account who you want to play with. They will be notified through their Elk Live account and must accept before you can invite them (or they can invite you) to join in a session. So, having got acceptances from all the group, you can start a new session and invite them to join.

You can see three tabs across the top of the page: Backstage, Video and Mixer. Backstage is where you invite your friends to play — in Screen 1, Rob is the only one to have accepted so far. Video is where you can see the other participants in the session, Zoom-style.

The Mixer is so called for obvious reasons, and as you can see in Screen 2, Rob has now joined us. On the left is where you set up your inputs: mic, line, guitar and so forth. There's also an option for USB that I'll talk about later. I'm using both mic inputs on the Bridge, one for my mic and the other for the third member of the group, my wife Anne.

You remember the MIDI and Lightpipe I/O ports on the Bridge? Those don't do anything yet, but Elk Audio have plans for them. In future versions of the software you will be able to send MIDI data to session participants and *vice versa*. The Lightpipe ports will support both S/PDIF and ADAT; the latter is particularly interesting, as that would give you up to an additional eight channels if you want them.

The gain you select in the Inputs section will affect the volume level you are heard at by your colleagues. The right-hand panel, on the other hand, affects the volume you hear in your headphones. There is a fader for each participant, including yourself — if anyone is using more than one input there will be a fader for each — and you can set the

»

Elk Live

PROS

- Bridge hardware is elegant and easy to use.
- Elk Live service can also be used with your own audio interface.
- Subscription currently free.
- Excellent low-latency performance.

CONS

- Native app currently Mac-only.

SUMMARY

Elk Audio have developed some impressive hardware and software that achieves low-latency music making across the net as effectively as anything else I've tried, but with greater ease of use than I've seen elsewhere. Just as importantly, they have some very good ideas about where they want to take the technology next.

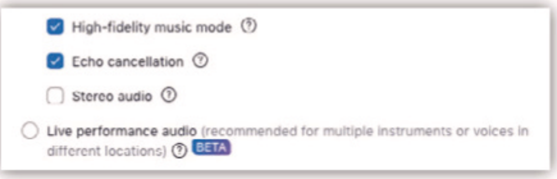
Zooming Ahead

While upgrading my Zoom client to version 5.14.0, I noticed among the audio preferences something I'd not seen before. It was an option called Live Performance Audio, marked prominently as a beta feature. Experiment showed that it can't be active at the same time as Original Sound for Musicians: when you select one, the other is deselected.

It had a tooltip which read as follows: "Reduces audio lag during live performances when musicians are on separate Zoom clients. Requires high-speed Internet or call quality will be significantly reduced." Further study revealed it needed more than that. In fact, the requirements are the same as those normally needed to support low-latency music making: a cabled rather than Wi-Fi Internet connection, and echo cancellation turned off, forcing the use of headphones rather than speakers. Further study with the help of uncle Google revealed that Zoom were hoping to achieve latency of 30-50 ms (the same sort of range Jamulus aims for), and also that all participants in a meeting had to have Live Performance Audio turned on or else it wouldn't work for anybody.

We decided to give it a try. Unfortunately Rob got a huge delay of a half second or so on his own voice, which, if ever you've tried it, makes conversation difficult and singing impossible. It only happened when Live Performance Audio was on and we never got to the bottom of that, so we weren't able to test it properly. Subsequently Anne and I tried it at home, each on a different computer, and it worked beautifully. We were able to sing close harmony together with ease. However, since we were on the same local network it would have been more disturbing if we hadn't had good latency. Of course the video wasn't synchronised, but that was little surprise and no problem.

Either way, it is clear that Zoom, who have a history of trying to cater for the needs of musicians, are seriously trying to get low-latency music-making implemented in their client for those who want it, and this must be welcomed. Of course Zoom isn't free and you wouldn't subscribe just for this facility, but if you or a fellow band member does have a paid-for Zoom account, this is well worth looking into.



☑ High-fidelity music mode ⓘ
☑ Echo cancellation ⓘ
☐ Stereo audio ⓘ
○ Live performance audio (recommended for multiple instruments or voices in different locations) ⓘ BETA

The new Live Performance Audio setting in Zoom.

relative levels of all the participants to give you the monitor mix that works for you; others are not affected by the settings here, as they have their own faders.

There are two headphone sockets on the Bridge, but only one works at a time, which is a slight shame since I and Anne both need headphones. However, that headphone output is more than adequate to support two sets of phones via a simple splitter cable, and so we were off.

The first time Rob and I tried Elk Live, we used mobile phones so we could talk to each other while we got things working. We made all the obvious mistakes like forgetting to switch on phantom power and not noticing that the master volume was down at zero, but once we'd got past those we were up and running in what was a remarkably straightforward and trouble-free link-up for nearly an hour until we decided to stop.

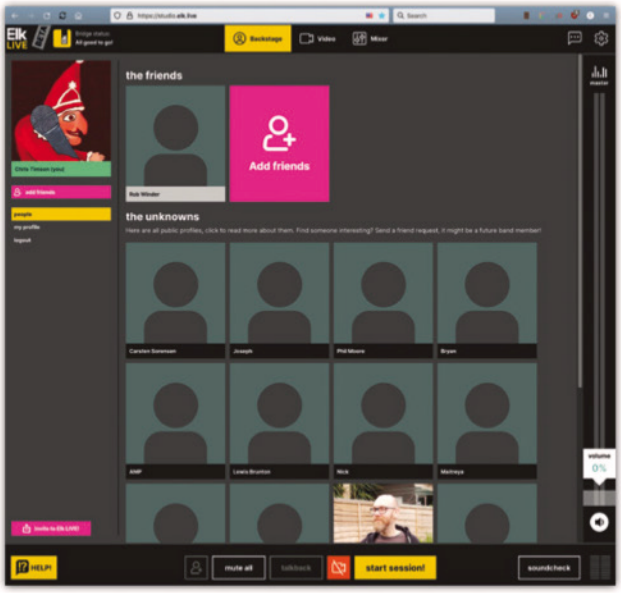
Live & Direct

And now the big question: what was the quality like? Well, it was excellent; virtually free of the pops, clicks and bubbling mud that sometimes afflicted us with Jamulus (though, in fairness, never enough to stop us singing), and latency was very low. We could sing in harmony together with confidence. We had video, though of course it was not sync'ed. There is an option to sync sound and video, but Elk Audio recommend you don't use it for music because of the high overheads it has. There's also a Zoom-style chat facility, which we didn't use much, though I can see it being pretty essential for anyone who's not using a mic.

In Screen 3 you can see a big red X at the bottom. Clicking it brings the session to an abrupt halt, something we did accidentally on one occasion — an "Are you sure?" prompt might be useful here.

It is possible to connect the Bridge to your computer via USB. This is in addition to the network link and does not replace it. If you do this, the Bridge appears as a class-compliant two-in/two-out audio interface, which is still controlled from the web interface. In its current implementation it is very limited. For example, if you want to output sound via the Bridge, you have to set both of its inputs to USB, so you can't use mics or instruments at the same time as USB input. But it does give you a way of recording what you can hear in your headphones. You need a DAW for this (it can be something as simple as Audacity) and you select the Bridge to be the stereo input to the DAW. You can then record the session as a stereo track.

That, at least, was the position up to the beginning of April. And then (flourish of trumpets please) everything changed. Elk Audio released a native software package they call Elk Live. This is a beta release, and currently runs only on macOS from version 11 onwards. It is standalone software and doesn't require the Bridge hardware, although if there is a Bridge on the network, Elk Live will find it and use it. Otherwise it will locate the audio interface you are using and treat it as if it were a Bridge, ie. a simple two-in/two-out interface (this, I understand, is a temporary restriction and will be lifted in a future version). Simultaneously, Elk Audio made the subscription element free, at least for the time being.



Screen 1: The Elk Live browser-based app's splash screen, where you can find friends and invite them to your session.

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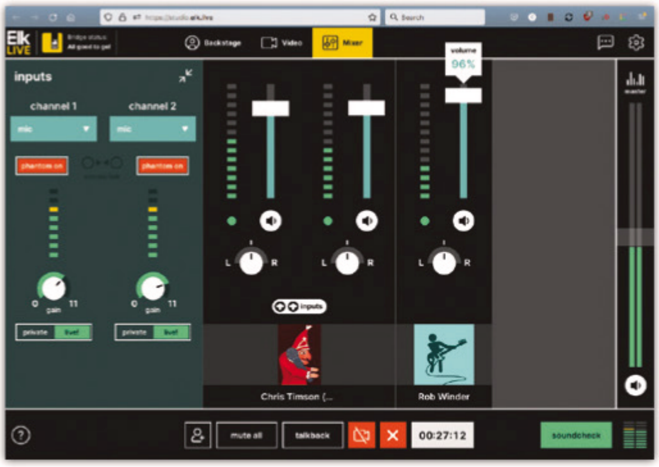
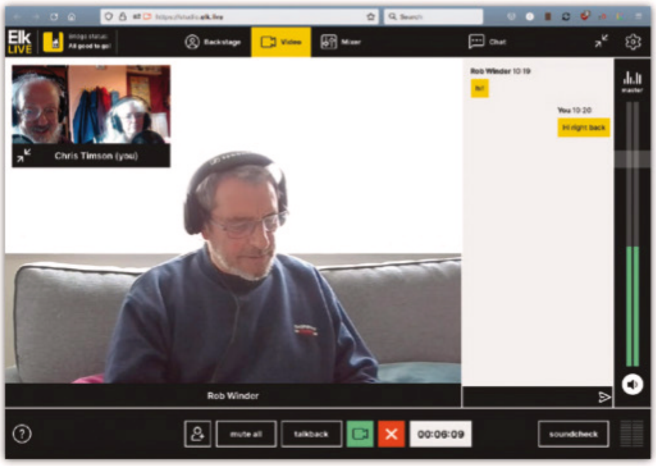


Screen 2: The Mixer page, with input levels on the left and return faders on the right.



Rob Winder 10:19
hi!
You 10:20
Hi right back

Screen 3: Elk Live incorporates Zoom-style video and text chat.

My colleague Rob uses a Windows machine, and so cannot use the native app. This gave us the opportunity to see whether the app and the Bridge could communicate successfully across the Internet. On starting the native app I was asked to log in, then to choose my interface, and after that the experience was almost identical to the web version with the Bridge, and completely trouble-free. The sound quality was as good, too.

One issue I encountered is that trying to run a DAW (Logic in my case) at the same time causes all sorts of confusion, and so for now you can't record a session at the native end. This is apparently a known issue and the only one we hit during our use of it.

The Future

So what we have here right now is some very nice hardware and software enabling musicians to make music together across the Internet. But Elk Audio are not alone in this; there are alternatives already available, such as the open-source system Jamulus I use regularly, and people like Zoom are beginning to take an interest (see box). But what I think makes Elk Audio particularly interesting is their plans for the future, given the firm platform they have built to start from. I had a long conversation with Björn Ehlers of the company, and here's a summary of where Elk Audio hope to take this technology.

The current free subscriptions are, as I've said, temporary, but Björn says they will remain free for at least two months (this at the beginning of April) while the company decide what pricing model they want to adopt for the long term. It will definitely be a 'freemium' model, though Björn was very certain that the free tier will be capable of worthwhile use, ie. doing the basic thing of playing together over the net. The paid tier would have some of the more advanced features discussed below. The balance between free and paid is very much under discussion right now. They view the free period as a chance to learn more about how people want to use the system, and for more people to learn about Elk Live. That is welcome news, as at the beginning I was uneasy about the cost for everyone of forking out £13 a month each (and that on top of the £300 for the Bridge).

The native app is also getting a lot of work, of course, what with producing a Windows version and extending the functionality to match the Bridge and beyond, but what they're also working on — and hope to have ready in a couple of months — is a VST plug-in that can run inside a DAW and act as a link to the native app. Once this is available, you should be able to play tracks from your DAW into the Elk Live session, where they could be heard by all the participants. It should also be possible to record whatever comes out of the Elk Live session back into your DAW, giving the facility to overdub parts over the Internet. Future development beyond that would include the plug-in being able to handle multi-channel output from the native app so each participant's output could be routed to a separate channel in the DAW. Elk Audio regard the VST as a priority and I can see how valuable this could be in many ways.

Longer-term, they're thinking of ways to synchronise video after the event using time stamps to support making videos for streaming and distribution. Further still, they're thinking about online live event staging with an audience watching and listening to the session — not just for concerts, of course, but masterclasses and other educational uses.

Big players such as Zoom are also getting into the low-latency act, and Björn welcomed this. The biggest challenge in making their proposition more widely known, he said, was simply convincing people that low-latency Internet audio is even possible! His view is that this is an underdeveloped area of the Internet and that the more people are involved in developing low-latency applications, the more it will be taken seriously, and the more opportunities there will be for companies like Elk Audio.

Now that the native app is out there and working well and being developed further, I had to ask Björn if there is still a future for the Bridge. As he says himself, you should get the same performance from the app as you do from the hardware. His reply was that, firstly, you can use the Bridge with devices such as iPads, whereas the native app needs a reasonably capable computer, and second, there are people who don't want to get involved with audio interfaces and stuff but would rather just plug a box into their network and get on and play. But he did accept that the main focus in future would be the native app.

At the end, Björn said that the basic message he was trying to get across was that Elk Live is free to download and free to use, and so anyone interested can give it a try. ■■■

£ Bridge £300 including VAT. Elk Live subscription currently free.
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Eve Audio SC2070

Active Monitors

Eve's new compact monitors borrow more than a little tech from their premium designs.

PAUL WHITE

While it might look familiar from the outside, Eve's new two-way active SC2070 monitor incorporates some significant advances over the company's previous models. In this case, Eve have allowed for a higher cost compared with their standard two-way range to enable them to produce a really high-quality design in a very compact footprint. Priced individually rather than in pairs, the SC2070 features the familiar folded-ribbon tweeter, aka Air Motion Transformer, but the RS7 version used here is the largest Eve have ever put in a two-way monitor. Its diaphragm area is 1.5 times bigger than those of Eve's other two-way models, and only slightly smaller than the AMT RS6 used in Eve's large SC3010 and SC3012 monitors. This crosses over at 1800Hz, which is unusually low, with the aim of providing midrange coverage equivalent to a three-way design.

The pleated construction of the AMT's diaphragm, which is located in a strong magnetic field, presents a much larger surface area than a flat ribbon, and the air motion is generated by adjacent folds alternately attracting or repeling each other as an audio signal current is passed through the conductive metal coating of a very light polyethylene terephthalate (PET) membrane. This is in contrast to a conventional ribbon tweeter, where the whole ribbon moves backwards and forwards. This motion actually occurs over a very small distance, but the folded structure still moves more air than a conventional tweeter of the same diameter.

Handling the lows is a 6.5-inch SilverCone woofer, also used in Eve's SC3070 and SC4070 monitors. This employs a copper-cap magnet system and a 1.5-inch voice coil. Its cone has a light but stiff honeycomb construction for low distortion, and is supported in a roll surround. Power comes from a pair of Class-D amplifiers with a combined power of 250 Watts and, as with other Eve monitors, the analogue input signal is converted to digital so that a DSP can provide additional optimisations as well as handling crossover, limiter and protection duties.

When the speakers are powered up, the volume level rises slowly to give the operator time to turn them down if they have been set too loud. The maximum input level can be switched between +7dBu and +22dBu to optimise the A-D converter resolution, and conversion is carried out at 24-bit, 192kHz using Burr-Brown converters. Analogue inputs are on unbalanced RCA phonos and balanced/unbalanced XLR; there's no quarter-inch jack provision.

SC2070
eve audio

The cabinet, which measures 230 x 345 x 304mm and is machined from MDF, has bevelled baffle edges to minimise cabinet edge diffraction and is recognisably an Eve design, with a satin-finished baffle and a vinyl foil finish to the cabinet top, back and sides. The connections and mains inlet are on the rear panel, alongside a switch to select between 115V and 230V operation.

With any ported speaker design there's a trade-off between bass overhang and low-frequency extension, as well as the risk of air movement noise from the port. In this design the slot-shaped port is at the rear of the cabinet, which reduces the impact of any residual port noise, and the port is larger than usual, with a rounded lower edge to minimise port distortion. A consideration with rear-ported monitors is that they have to be mounted a little distance away from the wall behind them, otherwise the proximity of the wall will compromise the port action.

In familiar Eve fashion, all the level, mute and EQ adjustments are made using a single 'turn and press' encoder surrounded by an LED ring that displays values. Thoughtfully, the brightness of the LED ring can be set to bright or dim to suit the ambient lighting in the studio, and there are two options for how the LED ring shows the current value: ring or dot. The EQ options comprise a low shelf (200Hz), mid EQ (1kHz) and high shelf (3kHz), and it is possible to lock volume and/or filter settings using the rear-panel DIP switches to prevent inadvertent adjustments. There's also a Desk filter, which has different functions depending on whether it is turned up or down. When turned down it works as a 170Hz notch filter to counter typical desk reflections, but turned up it provides an 80Hz bell-shaped boost that adds a little low-end punch.

The specifications for the speaker are impressive, with a free-field frequency response of 38Hz-25kHz (-3dB). A maximum SPL of 116dB is quoted, which is more than enough for use at close quarters, and Eve's response plot shows the frequency curve to be nominally flat when no filtering is applied.

In Use

I set up the SC2070s a metre or so from the front wall in my studio, which has enough

Eve Audio SC2070

£1299 each

PROS

- Smooth but detailed, non-fatiguing sound.
- Impressive bass extension.
- Equally impressive high-frequency reach.
- Practical EQ adjustments for room optimisation.

CONS

- No obvious cons other than a small amount of bass 'overhang', which is inevitable in all ported designs.

SUMMARY

The SC2070 is a highly capable monitor in a compact, two-way format.

acoustic treatment to give reliable results with my existing monitors. At typical monitoring levels the speakers sounded impressively smooth and solid even before I got around to tweaking the EQ settings. The low end reaches down rather further than you'd expect from what is a fairly compact two-way speaker, and though there's a little low-frequency overhang due to the use of a reflex port, it is certainly nothing that would worry me. That AMT tweeter really earns its keep here, as the highs sound smooth and effortless, reaching up beyond what most monitors can manage. In my room, the mids sounded very slightly recessed with the controls set flat, but that mid EQ came to the rescue, allowing me to add just a hint of 1kHz lift. The low end was also trimmed back by a dB or two to compensate for the room characteristics. Once dialled in, the monitors sounded extremely natural and I was comfortable using them for extended periods with no sense of fatigue.

While this is strictly a subjective rather than technical review, I've always found it useful to compare the overall tonal balance of monitors with that of a decent pair of open-backed headphones. Obviously the way the stereo image is presented differs between headphones and speakers, but the comparison is useful not only for checking the tonal balance but also for examining how tight the bass end sounds. Open-backed headphones don't suffer from port overhang so you get a fair sense of how something like a very dry kick drum is presented, and there are also no crossover issues to worry about.

Once I'd made the necessary EQ tweaks, the overall balance from the monitors came very close to that of my headphones, and while no ported speaker is entirely free from bass overhang, the SC2070s managed to sound well controlled and punchy with a lot of depth. Having that mid EQ control and separate Desk EQ really helps in getting the speakers optimised for your room. Voices came across sounding very natural, and those smooth highs translate to a great sense of detail without ever sounding aggressive. I also found that the stereo image presented a useful sense of depth as well as width, which is a trait usually associated with well-designed speaker.

In summary then, these monitors share the Eve family sound but with extra finesse when compared with their more affordable models. Even so, they are still priced within reach of the serious private studio owner as well as professionals.

£ £1299 per speaker, including VAT.
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PRE-73 DLX PREMIER
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MIC
Off
LINE/MIC
Power -10dBu +4dBu +14dBu Clip
80Hz 160Hz 300Hz 50Hz OFF
-7dB -14dB -21dB -28dB OFF
3dB OFF 6dB
INV
HIGHPASS
OUTPUT
OUTPUT PAD
AIR EQ
PHASE
PRE-73 DLX MKII
PROJECT

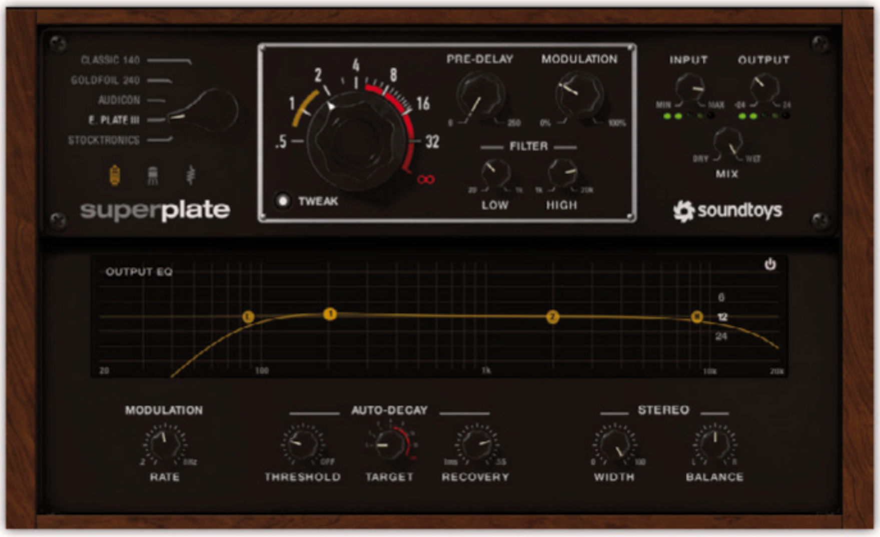
- Vintage style electronics, no surface mount components.
- Mostly soldered, connector-free internal design for the purest and most reliable signal path (PV = Premier version).
- UK-made Carnhill mic input and line output transformers and hp inductor (PV).
- Traditional 23-position 1073-style gain switch that is also used to select the mic or the line input jack.
- Combo XLR/TRS mic and line input jacks, separate output XLR and TRS jacks.
- Maximum gain on the mic input is 80 dB, Line input gain range: -20 to + 10 dB.
- Switchable impedance on the mic input, 1200 or 300 ohms.
- An active instrument input for guitar and bass that passes the signal through the mic input transformer, with selectable input impedance, hi-Z or 100 kohm.
- The phantom power phantom power resistors are matched to within 0.1% (PV).
- Selectable 1073-style inductor based highpass filter at 50, 80, 160 or 300 Hz.
- Selectable two position AIR EQ with a boost of about 3 or 6 dB at around 30 kHz.
- Switchable insert jack for inserting EQ´s and other units.
- An output pad allows the user to add character by attenuating the output signal after the output transformer by -7, -14, -21 or -28 dB for overdriving the output stage and the output transformer to a suitable level with the pad.
- Tantalum and polystyrene capacitors (PV) in the traditional positions.
- Switches on the back panel for 600 ohm output termination and ground lift.
- Relays are used to control internal switching for phantom power, mic low-Z, insert and DI for shorter circuit board signal paths.
- Circuit board star grounding scheme.
- Power supply with separate regulators for the audio circuit and the relay and LED circuit. External power supply to avoid interaction with the audio circuits and transformers.

Soundtoys SuperPlate

Reverb Plug-in

A new Soundtoys plug-in is always an event, and SuperPlate is their first release in over five years. Just how super is it?

SAM INGLIS



SuperPlate, showing Auto-Decay in action: the yellow bar around the main Decay Time control indicates that Auto-Decay has shortened the decay time in response to an input above the threshold.

Way back in 2017, Soundtoys introduced a plug-in called Little Plate. Like the other 'little' plug-ins in their range, it was initially free, and intended mainly to whet the appetite for a forthcoming full-fat version. However, Soundtoys have taken their time to get the latter right — and during that time, Little Plate has become a hugely popular plug-in in its own right.

The 'full fat' version was finally announced at this year's NAMM Show, and on the face of it, SuperPlate looks like a fairly natural extension of Little Plate, with the main additions being a selection of different plate emulations and a much-needed pre-delay option. Don't be fooled, though: there's a lot more to SuperPlate than meets the eye.

SuperPlate is sold separately and as part of the Soundtoys 5.4 bundle. It's available in all the major native formats, and is authorised using the iLok system. Unlike Little Plate, SuperPlate can also be used as a module within Soundtoys' EffectRack plug-in chainer. This means that if, for example, you want to use their classic EchoBoy delay plug-in to emulate a tape delay feeding into SuperPlate, you can save the entire chain as a single, transferrable plug-in preset.

On A Plate

Little Plate was inspired by the classic EMT 140, and numerous examples of the original were tested during its making, but the aim was never to make a 'warts and all' recreation. Rather, Soundtoys tried to capture the essential character of a particularly good example, and implement it so that it was usable beyond its usual range of decay times. In SuperPlate, they've extended the same approach to four more historic plate reverb models. The EMT 240 was a later design that used stretched gold foil in place of the 140's steel plate, allowing it to be made much more compact, and giving it a noticeably different sound. Two more models are based around locally popular American plates from the 1970s and '80s, while the fifth is based on the rare Swedish Stocktronics RX4000. All five models offer the same expanded range of decay times, from half a second to infinity (RT60 at 500Hz), and the new pre-delay is variable up to 250ms for special effects.

In order to create reverberation, a physical plate needs to be stimulated, and this is done by amplifying the source signal to feed a driver coupled to the plate. In the early EMT reverbs, the amplifier was a valve design, while later models used a solid-state amp. Each has a characteristic sound, especially when overdriven. (Unusually, nearly all engineers seem to agree that the solid-state version is better, perhaps partly because it's less noisy.) Soundtoys have modelled them both, as well as offering a clean alternative. Feeding a hotter level into the plug-in or boosting the input gain control gradually increases the amount of compression and colour coming from the modelled amplifier circuitry, if enabled.

Duck Tails

Like most of Soundtoys' grown-up plug-ins, SuperPlate has a Tweak button that opens a pane of extra controls. These include an additional two-band equaliser, which is hard-wired at the output of the plug-in. Although it's displayed on the same graph, the high-pass filter inherited from Little Plate is on the input side; here, it is joined by an extremely useful low-pass filter, and both can be switched between 6, 12 and 24 dB/octave slopes. The Tweak pane also allows you to alter the rate of the modulation and adjust the stereo width (see box), but the most intriguing element is something Soundtoys call Auto-Decay.

Most of us are probably familiar with the concept of ducking, whereby a compressor is placed after the reverb and triggered from the dry signal feeding the reverb. If you get the settings right, this can allow the reverb to be pushed down in level while, say, a vocal line is being sung, before rising up at the end of each phrase. Ducking can help to

Soundtoys SuperPlate
$149

PROS
- Sounds superb, and retains the character of a real plate whilst being much more versatile.
- Beautifully thought-out control set makes it a pleasure to use.
- Auto-Decay is a clever and effective alternative to ducking.

CONS
- None.

SUMMARY
Was SuperPlate worth the wait? You bet. This is a truly excellent reverb plug-in that sounds great on practically everything.

maintain intelligibility and keep the vocal at the front of the mix, whilst also making it sound lush and rich.

Auto-Decay is an alternative way of achieving a similar goal, by modulating the duration of the reverb instead of its amplitude. You set a Threshold and a Target decay time; when the input signal exceeds the threshold, the reverb decay time is turned down from its main setting to the Target value. Once the signal falls below the threshold again, decay time reverts to the normal value, at a rate determined by the Recovery dial.

It's a clever and highly effective implementation. In a vocal context, Auto-Decay can achieve much the same thing as ducking, and it's very easy to set up, but it has novel applications too. For example, if you set a slow Recovery time on a snare sound, you can create a fantastic non-linear reverb that blooms after the drum has been hit. It's enough to make me wish that SuperPlate could accept an external side-chain signal to trigger Auto-Decay, because that could be a lot of fun, especially with otherwise-infinite reverbs. The experimentally minded might also mourn the fact that Auto-Decay can't be used in reverse, to lengthen rather than shorten the decay time.

Hot Metal

In the main, though, and infinite reverbs notwithstanding, SuperPlate isn't primarily an experimental tool. It's intended to be a bloody good reverb for mixing — and that's exactly what it is. As with so many other Soundtoys plug-ins, the genius lies in the way the designers have boiled everything down to present exactly the right balance of versatility, ease of use and sound quality. There are few enough parameters that operation becomes instinctive after half an hour or so, but there are sufficiently many that you rarely find yourself wishing for control over something that is fixed. Moving any control has clearly audible results, yet none of them has a narrow sweet spot. And the reverb always has that complex, dense 'plate-y' quality, even when you're pushing the decay times way beyond the capabilities of a real plate.

The EMT 140 is still probably king of the hill in terms of producing a sound that works for absolutely anything, but the other plate models are very usable, and sometimes preferable especially on sources such as snare drums. The 240 has a boxier, more prosaic quality that can be just the ticket where you don't need the full dreamy floatiness of the 140, while the Audicon has a brash and slightly 'tiled bathroom' tone, with a strong upper-midrange emphasis. The Ecoplate III is warm and solid and often makes a good meat-and-potatoes alternative to the 140, while the Stocktronics plate is distinctively bright and splashy. And in every case, the amp modelling is a surprisingly intuitive and powerful tool for shaping the sound.

Super Stereo

Most hardware plate reverbs were either mono or mono-in/stereo-out devices, and Soundtoys' earlier Little Plate plug-in took the latter approach: it always summed the input to mono, even if used on a stereo aux or track, so panning your reverb sends made no difference to the end result. SuperPlate is a stereo-in, stereo-out plug-in, and the Tweak panel includes both Width and Balance controls. With the Width set to its default 100 percent, SuperPlate actually seems to behave more like a dual-mono processor, in that a hard-panned input only generates output on one side. So, if you want to use it to add some stereo 'spread' to a hard-panned source, you'll need either to pan the reverb send differently from the source, or reduce rather than increase the Width. Or, of course, you could use a mono-to-stereo instance in DAWs that allow it.

A quick straw poll of the other plate reverb plug-ins in my folder suggested that Rare Signals' Transatlantic Plate behaves the same way as SuperPlate, whilst EMT 140 emulations from Arturia and Universal Audio are always mono-in/stereo-out, and Liquidsonics' Lustrous Plates is somewhere in between.

I frequently use half a dozen or more different reverbs in a mix. Sometimes I'll even combine two or three on a single source in order to get the effect I'm looking for. And although one or two of them might be plate emulations, I'd usually lean pretty heavily on other reverb types too. When I was testing SuperPlate, I swapped it in for all the various reverbs in an existing mix in turn — and in every single case, it stayed. If you think mix engineers cannot live by plate reverb alone, an hour or two with SuperPlate might just change your mind. This is a plug-in that lives up to its name and more.

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Jacob Morris. This photo was taken in the studio he helped build for artist David Crowder.

INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

Jacob 'Biz' Morris

Today's TikTok-driven music market demands authenticity, as Jacob Morris found when he mixed David Kushner's 'Daylight'.

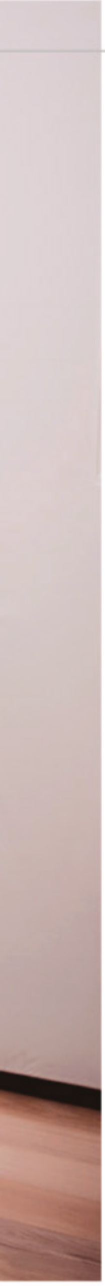
PAUL TINGEN

While it's common knowledge that many new artists today break through via TikTok, few people seem to be aware of the scale of the changes the platform has brought to the music industry. For example, in 2020 alone, 70 artists that went viral on TikTok went on to sign deals with major labels.

TikTok has also greatly impacted music itself. It is the domain of short, home-made videos, and an impromptu, amateur quality is often seen as an asset rather than a drawback. The fact that pop songs are becoming shorter and shorter is in part due to TikTok, and the question has been asked when the first 15-second pop hit will be upon us. In addition, the amateur approach has begun to transfer to the music.

Artists now float bits of unfinished songs on TikTok to see if they get any traction, accompanied by what are called 'evergreen' pre-saves, which allow artists to promote future music without a release date or even a song title. If and when these rough demos get traction, the imperfections of the demo often inform the end product.

American singer-songwriter David Kushner is a case in point. Reportedly without having a music career in mind, he released the guitar song 'Miserable Man' on TikTok in January 2022. It went viral, and eventually became a major hit in many countries. The same happened, to



a lesser degree, with his second single, ‘Mr Forgettable’. Both songs sound like embellished guitar/vocal demos, and were created on Kushner’s iPhone.

The two singles found their way onto Kushner’s debut EP *Footprints I Found*, and by the end of 2022, he had amassed half a billion streams and close to one million followers on Spotify alone.

Earlier this year, Kushner dropped a snippet of a new demo on TikTok, and pushed it with the meme “You look happier; what happened.” It went viral, and the rush was on to finish the track. Produced by Rob Kirwan, ‘Daylight’ was officially released on April 14th, and became Kushner’s most successful release to date. By the time this article went to print, it was at its third week at number two on the UK charts, and had reached number one in several other countries.

Don’t Fix It In The Mix

‘Daylight’ makes a strong feature of its imperfections, with a piano part that has pedal sounds and other noises, and distortion on the vocal. When Atlanta mixer Jacob ‘Biz’ Morris received the song for mixdown last March, he initially tried to fix some of these artefacts, assuming they were mistakes.

“The second chorus gang vocals sounded really distorted. You can hear the room and the mic capsule distortion — he’s really screaming in the thing. I did my best to clean it up, but received a mix note from David that said, ‘No, no, let’s keep it dirty.’ That chorus is everything in that song, and he wanted it really dirty and vibey.

“I’m not on TikTok, so I had not heard the initial version of the song that had blown up on that, but when someone played it to me, I was like, ‘Oh, now I know where he’s trying to go.’ It was really grungey. People had been listening to that on TikTok, so the final record had to match that. David wanted the record to fit the vibe that everyone had already heard.

“Because the song needed to be dirty, vibey and emotional, I stopped trying to clean things up, which would have taken the song too much into a pop direction. In fact, I dirtied up those vocals even more with the Soundtoys Decapitator. I also left all the extraneous noises on the piano. Once again, the rawness is what makes that song special.”

Top Of The Class

Morris first started to work with Kushner earlier in 2023. “David’s manager had been following me for a while, and reached out to me to do some work for another artist, and he then asked me whether I would master David’s single ‘Elk Grove’. David loved what I did, and when they knew ‘Daylight’ was coming, his manager told me they wanted me to mix the song.”

It was an interesting choice, as Morris is better known for rap and pop music. He won two Grammy Awards, and was nominated while still at college, for his work with the Christian rapper Lecrae.

“I’m from Dallas, Texas, and always loved audio,” remembers Morris, “and I was fascinated by how music was created. How the people behind the scenes got a good-quality sound really interested me, and I knew from a young age that I wanted to be a recording engineer and a mixer. My mum was very supportive, and bought me gear, and by the time I was 14, I had an entire Pro Tools HD rig! I took it very serious, and went to the Conservatory of Recording Arts and Sciences in Phoenix.”

It was while Morris was a student at the Conservatory that he had an engineering credit on Lecrae’s album *Rehab* (2010), which was nominated for a Grammy Award for best Rock or Pop Gospel album. Morris also worked on Lecrae’s Grammy-winning *Gravity* album (2012), which also won a Dove Award, and there was another Grammy Award for Lecrae’s single ‘Messengers’ (2014).

“It was kind of funny to be learning in school from teachers doing their thing while I was the one going to the Grammy Awards!” remarks Morris. “After school I initially returned to Dallas, but soon moved to Atlanta, in 2013, because of my connection with Lecrae and Reach Records. I’ve been working in the studios in their headquarters since then. I do a lot of work for the label, but today I’d say that 90 percent of my work is with artists from outside the label.”

Morris’ recent credits include David Crowder — he also collaborated with Gavin Haverstick and the artist to build Crowder’s studio — as well as Social Club Misfits, Nobigdyl, Andy Mineo, Switchfoot, Lecrea, and many others. They are mostly Christian acts, something which presumably also fostered the connection with Kushner, who is vocal about his Christian faith.

Hit Machine

Almost all Morris’ work today consists of mixing and/or mastering, in both stereo and Atmos, in his room at Reach Records studios in Atlanta, with the help of several assistants, who use additional rooms. “I’m about 80 percent in the box,” explains Morris, “and regularly use outboard gear. My favourite pieces of outboard at the moment are the Tube-Tech CL-1B [*compressor*] and the Neve 1073 [*preamp and EQ*]. If I feel a vocal is missing something, I will often send it through the Tube-Tech and Neve, and I ran David Kushner’s vocals through them. I also have a Burl converter, and I regularly route my kicks through a Drawmer MX40 gate. It can give you a tone and real punch. I also have the dbx 160 [*compressor*], and I had the Dangerous compressor for a while, but I just sold that.

“I love the tone outboard can give you, and I think it’s a tone that you can’t get in the box. I also enjoy being able to touch gear with my hands. Just using the mouse doesn’t get my brain going, I don’t get in that flow state. You have endless plug-ins that have presets that you can just pick. You don’t have to work at getting a tone any more. But with outboard gear you have to know how to use it to be able to get it to sound right.

“The other thing about outboard is that it forces you to make decisions. Just print it and commit. I don’t even do recall sheets anymore. If something needs to be redone, I’ll restart from scratch. I don’t have a crazy amount of outboard, so sometimes I’ll go look for a piece of gear that I want to use on a particular song, and I’ll rent that in, or I’ll buy it, and use it and then I re-sell it. It drives my team

»

‘Daylight’

Written by David Kushner, Hayden Robert Hubers, Jeremy Fedryk & Josh Bruce Williams
Produced by Rob Kirwan

COMING IN HOT

Jacob Morris already mixes in Atmos and his new studio, shown in this mock-up, will be equipped with a Focal immersive monitoring rig.

mad! I have to keep things pretty slim, because I always have to buy three of each piece of gear, so my assistants can have the same.

"My monitors are the PMC TwoTwo8, and I also have some Yamaha NS10s, and the Trinnov speaker calibration system. My whole room is Atmos PMC. But I'm building a new studio next to my home at the moment, designed by Gavin Haverstick, and I'll be going back to Focal speakers. I used Focal Trios in the past. I tested ATCs and PMCs, because I was feeling like I was missing the midrange. But there's something about those Focals. I can listen to them for 12 hours, and my ears don't get fatigued. It's not that the PMCs are bad, but I just miss my Trios a lot.

"My third child was born the day I received the files for 'Daylight', and with three children now, I want to be closer to home. I want my workflow to fit in with my family, not the other way round. The studio is designed by Gavin and has loads of windows, and it will be fully set up for Atmos. I hope to be able to move in next month. We'll keep the four rooms at Reach, where my assistants will be able to continue working."

Be Prepared

"When people send me files, I'm dealing with Pro Tools sessions most of the time, and people on Ableton and Logic and so on send me audio files. 'Daylight' came in as a Pro Tools session. Although our baby was coming the same day I received the session, I said to my right-hand guy Connor, 'We gotta do it.' So one of my assistants immediately prepped the session for me. This means colour-coding, labelling, routing stuff and putting it into folders.

"Unless a track is like big pop or something, I do not want my assistants to clean up the audio tracks, because people may want the background breaths, or other things. For this reason I want to make those decisions myself. As I mentioned, on 'Daylight' there was a lot of noise on the piano, but the piano has a great tone, and the noises were part of the vibe. There actually was one part where David did want us to clean it up a little bit, but there was no way to do it, so we left it.

"My team uses a template, which is mostly routing and folders, and there also is a bunch of aux effect tracks that I can pull in at any time. They are Valhalla Room, VintageVerb and Plate [*reverbs*], some Soundtoys EchoBoy delays — eighth, quarter, half-note and slap — the Waves H-Delay and Doubler, and three different Audio Ease Altiverbs. Having all the routing in place and all the tools easily accessible means that you can mix much faster. My sessions are also organised in such a way that at the end of the mix, when the labels want the deliverables, we just hit go and the deliverables are printed automatically."

Morris repeatedly stresses the importance of being very organised before he starts mixing. "At any time the artist can make a request, and you have to be able to go do it immediately. I've seen a lot of people mess up just because they're not clean and organised. That could really hurt you as an engineer. Of course, what works for my team and me may not work for someone else. The brain of another person may work differently.

"I have the reference tracks right at the top, and after that everything is organised in folders. My sessions start with a bass folder, with all the bass tracks including synth bass, and then a drums folder. If

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GETMXD

A few months ago, Jacob Morris started a new online mixing service. GETMXD offers finished full stereo mixes for $500 per track, Atmos mixes for the same price, mastering for $125 and revisions for $30. For a Grammy Award-winning mixer, these prices seem almost too good to be true. Morris offers a peek under the bonnet.

"My goal is to help independent artists. They have great art, but they don't always have a great sound, partially because they can't afford to get mixing and mastering done properly. I wanted to break that barrier, and I built a system to make the workflow easier and more affordable. Of course, there's the value of my time, but if I can cut that down, I can cut the rate down.

"Five hundred dollars per mix is crazy, but we have created a whole system on the GETMXD site that streamlines the biggest issue, file management and communication. Sometimes it's me, sometimes it's my right-hand Connor who is doing these mixes. I also have two assistants, and we have a good team that works very well. I figured out a way to do it much faster, and it's not taking me much time. I would say in general we work extremely fast. We have the system down.

"Also, because of the success of 'Daylight', DMs are definitely going! We've been pushing all these requests to the site. So we're pretty busy, and some months we're doing 30-40 songs. David's has been one of the pieces, but for the rest it's word of mouth. I'm not big on social media. I don't try to be out there. If the work comes in it comes in, and if it doesn't, it doesn't."

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» there are live drums, I'll also have a live drums folder. Then there's a music folder, with all guitars and keyboards and so on. Underneath that there is an All Vox folder, a Mix Bus folder, which is where everything's going, then the Master track and finally a print track."

Respect The Rough

"With the session prepped, I start my mixing process by listening to the song a couple of times, trying to get an idea of where the artist wants to go with it. It's vitally important to know that, because what you get is where they're at. You have these artists who have already spent 12, 14, 16, even 48 hours crafting their mix. Ninety percent of the time, the music is already there, and I just need to do some volume changes and some master bus stuff to the music.

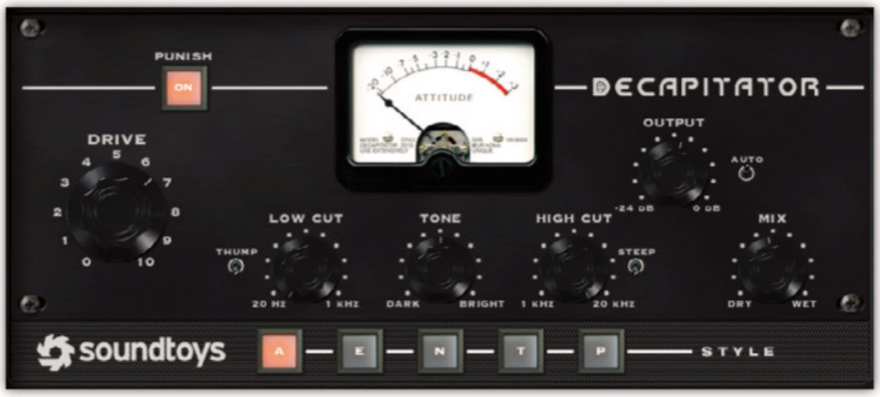
"In general, our job as a mixer is not to recreate the sound of a track, but just glue it and enhance it. It all depends on your relationship with the artist, but I think most of the time your job as a mixer is not to be a creative person. Your job is to read the vision of the artist and their song, and then execute it. Some mix engineers want to be creative, and so they spent four or five, six, seven hours on a mix, and I'm like, 'You're doing too much, just focus on what the artist wants to execute.'

"I don't try to be the most creative mixer, unless I know the artist. Usually, most of the mix work is already done, and I just take it and move it to the next level, where I get stuff to pop out where it needs to pop out. Working on vocals often is a large part of that, to get them to cut through the track. That can be a lot of work. But on the music side, it's often more straightforward.

"I always have the rough mix on a separate track at the top, so I can make sure I'm not straying too far from the feeling that they give. My assistants also make sure that the rough and the session sound the same before. I need the files that are coming in to 100 percent match the last bounce that they did. I need to be able to start where they ended. One client listened to his rough over 300 times! I cannot simply take a leftfield turn with my mix. Today, if I'm trying to fix things in a mix, I risk ruining it."

Rapid Turnaround

In the case of 'Daylight', Morris' process was slightly unusual, for a number of reasons. "The song was recorded the day before I got the session, so they were moving really fast, and they needed the mix, mastering, and the Atmos mix done in a couple of days. It was a really fast turnaround!

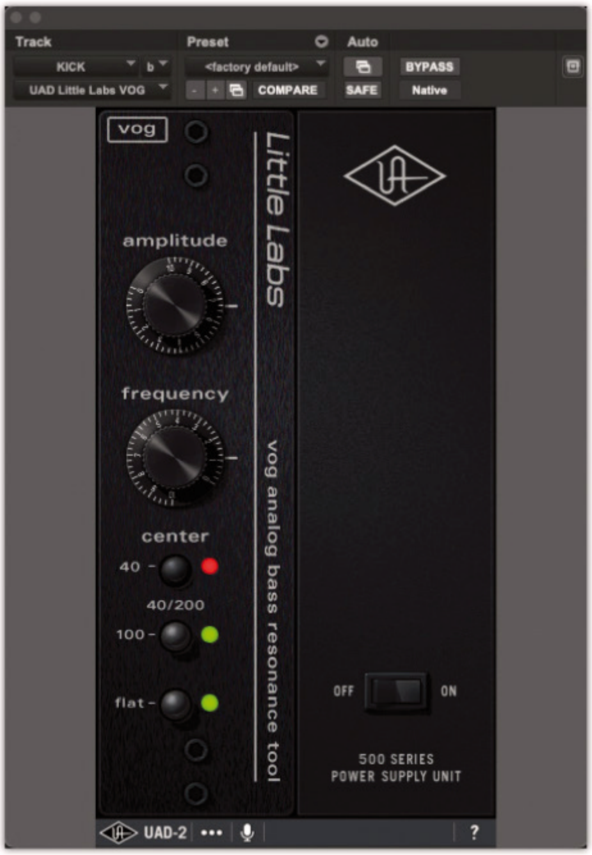


David Kushner's re-recorded vocals for the second verse were grunged up further with Soundtoys' Decapitator.

"I will normally have heard a song before I agree to mix it, to see if I feel I can add value to it, but in this case, I first heard the song when it came in. They were still in recording mode, and they re-cut David's lead vocals for the second verse while I was mixing, so I had to work them in while I was mixing. This meant that my entire process took three days instead of two.

"In general, once I have listened to the rough a couple of times, I'll listen to each of the folders and get a feel for what needs to be done and cleaned up in each folder. From there I decide where to start my mix process. With the 'Daylight' session, I started my mix with David's lead vocal. I soloed it and decided how to process it. His vocals are big, and I knew that they needed to sit on top. They're the most important element, so I needed the best version of that vocal.

"After that I worked on the piano, which is the second most important element of the track, and from there it was background vocals, and then drums, bass, and music. Of course, as I'm working, I'll be going back and forth between these elements and adjusting the processing on each."



The UAD emulation of Little Labs' Voice Of God bass enhancer was used to beef up the kick drum.

Getting The Treatment

"I used outboard on David's vocals, the 1073 and CL-1B, boosting a bit of high end with the 1073. After the vocals came back into the box, I de-essed them with the FabFilter Pro-DS, and then I hit them with the UAD LA-2A plug-in for even more compression. There already was reverb on many of the tracks, including on the vocals, but I added some of my own: on the vocals, the Valhalla Room, and some Altiverb as well. »

Rob Kirwan's studio is full of interesting analogue gear, including the Maestro Echoplex delay (top left) and Ekdahl Moisturizer spring reverb (behind the Echoplex) used on David Kushner's vocals.

Rob Kirwan: Producing 'Daylight'

Irish producer, mixer and engineer Rob Kirwan has worked with Hozier, PJ Harvey, Depeche Mode, Editors, Glasvegas and U2, and featured in our Inside Track series in March 2015, discussing his work on Hozier's mega hit 'Take Me To Church'. David Kushner is a big fan of Hozier, and the two artists have similar voices and musical directions, so it is not surprising that Kushner contacted Kirwan to help him with the production of a new song.

"David contacted me directly through my management," Kirwan recalls. "He really liked my work on the first Hozier album and wanted to see if any of that sonic mood and palette would be applicable to his new song. He sent me a very early and quite short version of what was to become 'Daylight'. Once I heard the chorus I knew that I could help bring a similar emotional journey and uplift to the song.

"David and I had a Zoom call and I proposed my vision for the song and he seemed very receptive. I asked for another demo version that incorporated some of the ideas we discussed, which David did brilliantly. We felt that the new demo was a real improvement, but the song needed some more work structurally and arrangement-wise, as well as some different vocal approaches. As we were only doing one song, we decided that it would be more beneficial to continue working remotely.

"David recorded some more stuff, like piano and vocals, in LA. I loved the squeaks on the piano, so keeping them was a must! I don't know what mic was used on his vocals, but I processed them through my Teletronix LA-2A compressor and Inward Connections Vac Rac EQ, and I also used my Knas Ekdahl Moisturizer spring reverb and the Echoplex on both his lead vocals and the backing vocals.

"I recorded guitars, synths, percussion, and did processing in my place. 'Daylight' was actually the first record I worked on in my new studio! It had literally been finished a week before and I was still covered in dust and plugging cables in the mornings, while LA slept.

"I felt that the lyrical content of David's words was quite dark, but also uplifting, so that informed the grittiness and the weight of the production. Grit stands for distortion, with varying degrees of subtlety, and weight for bottom end. For me, it was crucial that the song remained as honest as possible. David has an amazing ability to convey real emotion without being over earnest, which is a rare talent.

"Funnily enough, as I was finishing the song in my studio, my 14-year-old daughter was already listening to the early version on TikTok. It was a very strange feeling. I'm used to the music I work on taking months to come out, and this was so immediate. I can't say that the grittiness of the track came from imitating the TikTok version. I like dark and gritty in any case, and push most of my music and productions in that direction. My next release is the new PJ Harvey record!

"Biz got involved through Brent Shows, David's manager, and I thought he did an incredible job. He kept the emotional journey of the song, while improving the sonics. That describes pretty much 100 percent what I'm looking for from a mixer. He smashed it!"

Rob Kirwan in Assault & Battery Studios, London.

Atmos

"I have a separate session for the Atmos mix," says Jacob Morris. "It has all the stems and the multis [*individual tracks*]. Sometimes I use only the stems and I will pull in one multi if I need that one element. I don't just make a copy of the mix session for the Atmos mix, because I don't want all the processing in it. Instead I print everything, so everything is committed and the Atmos session has no plug-ins in it. Sometimes we have to send the Atmos sessions off to the label, and I want them to be able to open it."

»

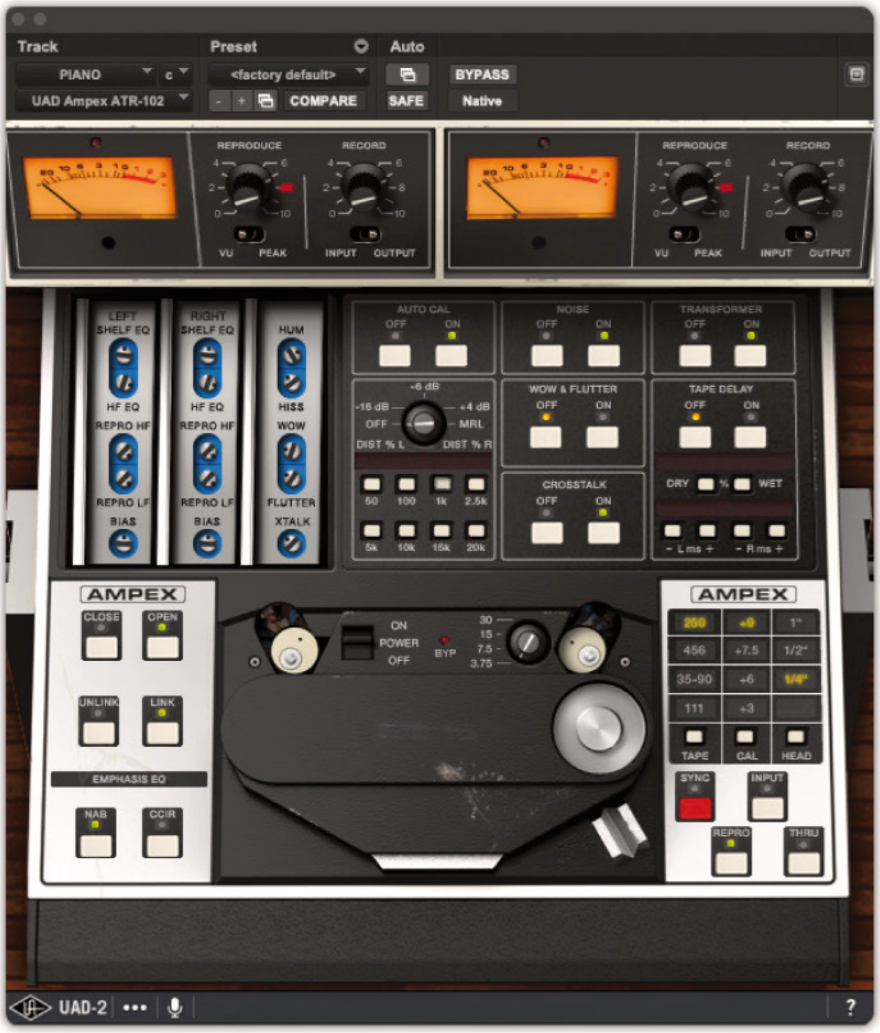
"The two new takes of the second verse had been recorded on Logic in the middle of the night, and were ready to be mixed in. However, that second verse, if you listen, sounds different. It's definitely more roomy. I sent them though my CL-1B and 1073 as well, and dirtied them up further with the Decapitator. I also adjusted the tone with two FabFilter Pro-Q 3 EQs. That EQ is such a good plug-in.

"I also used the Waves SSL G-channel plug-in, and the UAD LA-2A again, to make it fill up, and the compression brings out the dirt. By bringing out all that noise behind him and between his words, we got that vibe in that he was looking for. Normally I cut that stuff out! By the way, I did not run the background vocals through outboard, they were treated with all plug-ins doing really subtle things."

"The piano had quite a few treatments. I'm compressing it quite hard with the UAD Shadow Hills Mastering Compressor, and then I add the UAD Ampex ATR-102 and Decapitator, again to bring out that room tone out and the dirtiness. These are vibe moves. I'm also boosting the top end with the Pro-Q 3, and I also have the Valhalla VintageVerb on the piano.

"After that I went to drums and bass. That second half of the song where they come in is so big. I am compressing the overall drums with the FabFilter Pro-C 2 and the UAD Fatso, and doing a high-pass with the FabFilter Pro-Q 3. On the kicks I have the Pro-C2, and the Little Labs Voice Of God, which is really good. I am also using a lot of Decapitator on the overall drums, again for the vibe.

"In addition to the drums and stomps there's some bass that comes in on the third hook. They are a big thing. We used UAD Cambridge EQ riding some top on that, and also pushing up the bottom end. There's a very big synth bass too, to which I added Altiverb, to push it back a little bit. You don't want it up front. The main things in the track are the vocals and piano.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The piano was subject to multiple treatments to bring out the "dirtiness", including the UAD Ampex plug-in.

"There also are some subtle things, like acoustic guitar swells, but they are more of a feeling, and not very audible. I have the UAD Studer [*tape emulation*] on the Music folder as a whole, once again for vibe. Because of my hip-hop side the low end and drums are very important to me. I'm getting a lot of clients outside of hip-hop, and I think it's because they're wanting that hip-hop low end.

"On my mix bus I have Jaycen Joshua's The God Particle plug-in, and the FabFilter Pro-L2, for volume. The song was almost there, but because I knew I was also mastering this song, I did not do too much on the mix bus, and I simply took off the L2, and then printed the mix."

Finishing Touches

"I work in a separate mastering session. I have a mastering template, but the only plug-in that's already in it is the TC Electronic Clarity Meter. When I'm mastering an album, I have to make decisions about how the entire album is going to flow, but even then I don't feel like I can just open up a template with a whole bunch of plug-ins. The only thing that works is making the decision at the moment of mastering. Sometimes it's just a little bit of compression and limiting because the mix is already there, sometimes I do far more.

"In the case of 'Daylight', I'm using Oeksound Soothe 2 to clean up the low mids, the UAD Oxford Inflator to get a little more crunch overall, the FabFilter Pro-Q 3 and UAD Curve Bender EQs, and after that I'm slamming it with the FabFilter Pro-L2 for volume. I used to use the Dangerous [*outboard*] stuff when mastering, but I don't have that any more. Mastering this song was all in the box."

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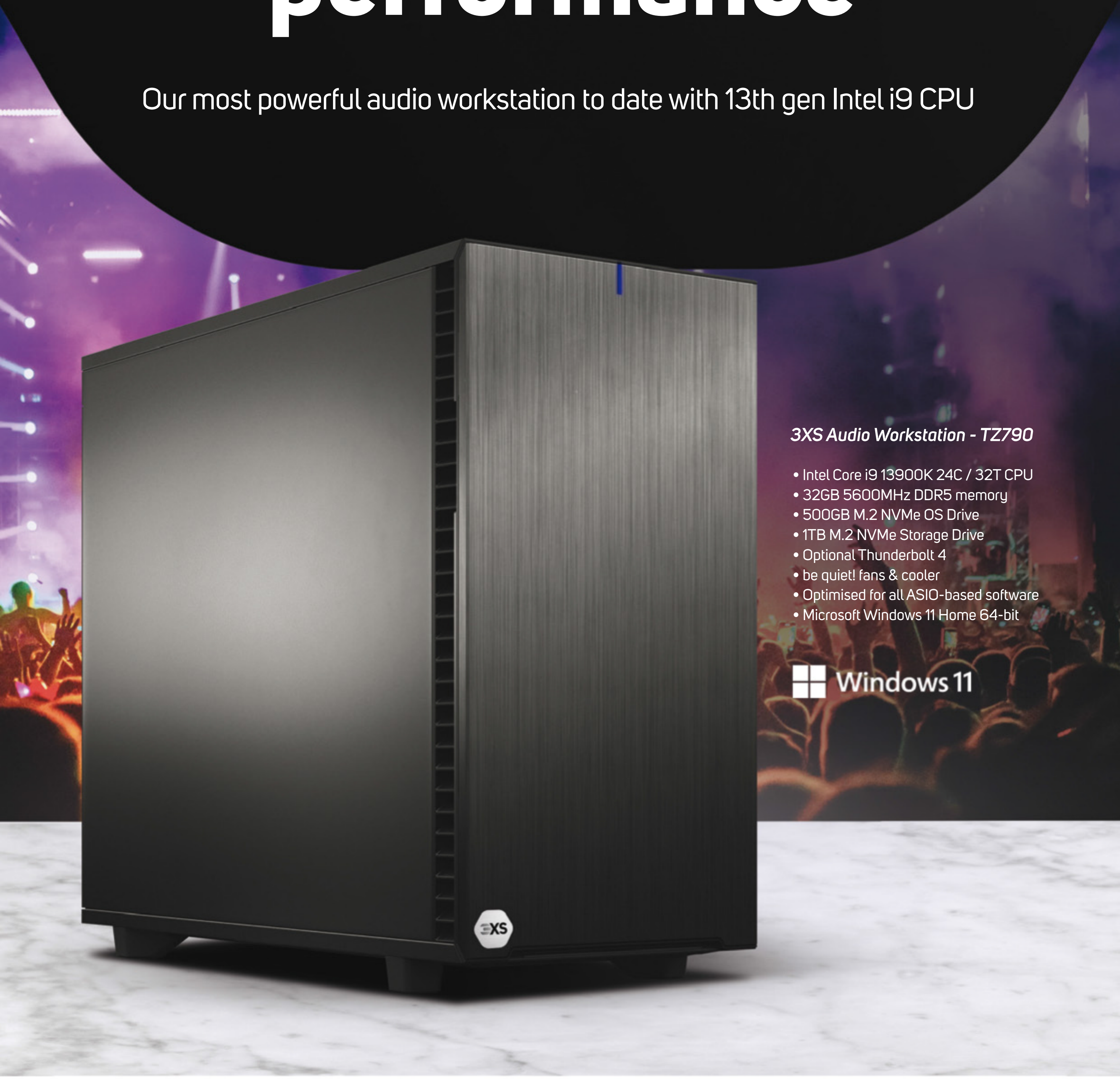

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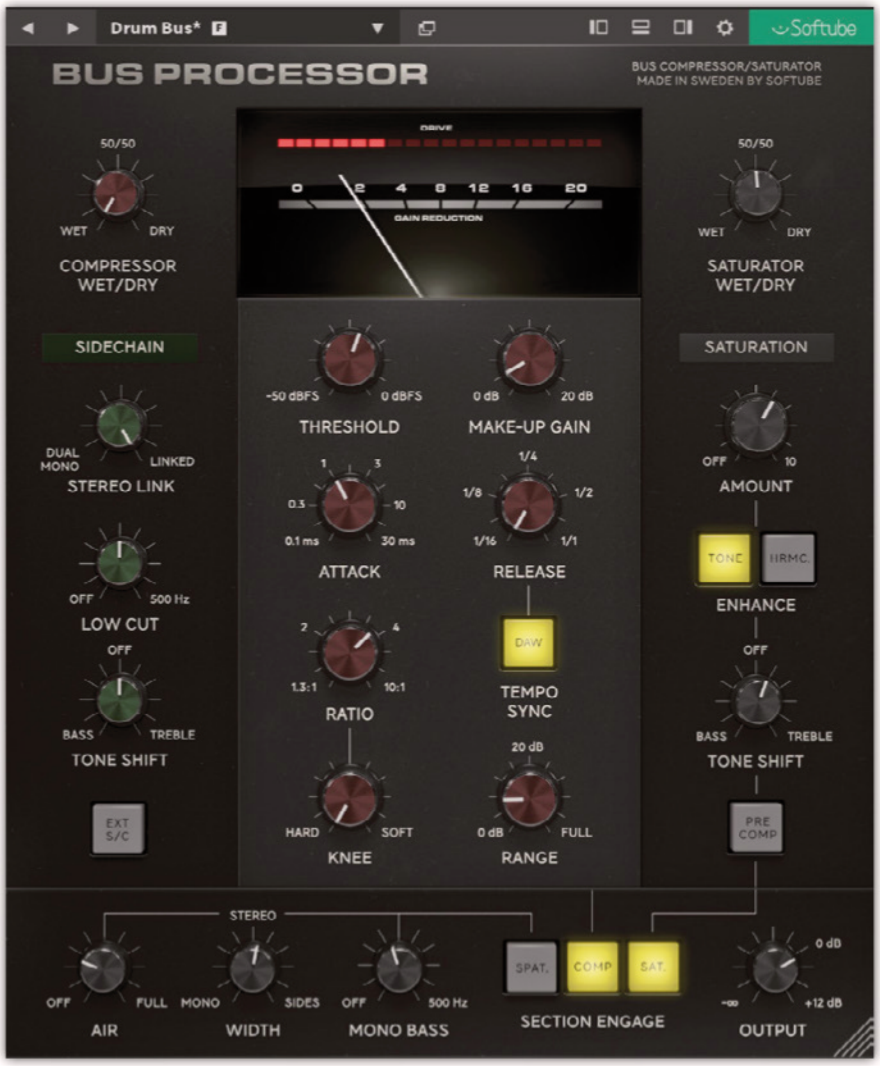
SCAN

Softube Bus Processor

Dynamics & Saturation Plug-in

As a traditional bus compressor, this plug-in sounds great — but there's much more to it than that!

JOHN WALDEN


Drum Bus*
Softube
BUS PROCESSOR
BUS COMPRESSOR/SATURATOR
MADE IN SWEDEN BY SOFTUBE
DRIVE
0 2 4 8 12 16 20
GAIN REDUCTION
50/50
WET DRY
COMPRESSOR WET/DRY
SIDECHAIN
DUAL MONO LINKED
STEREO LINK
OFF 500 Hz
LOW CUT
OFF
BASS TREBLE
TONE SHIFT
EXT S/C
-50 dBFS 0 dBFS
THRESHOLD
0 dB 20 dB
MAKE-UP GAIN
1 3 0.3 10 0.1 ms 30 ms
ATTACK
1/4 1/8 1/2 1/16 1/1
RELEASE
2 4 1.3:1 10:1
RATIO
DAW
TEMPO SYNC
HARD SOFT
KNEE
20 dB
0 dB FULL
RANGE
50/50
WET DRY
SATURATOR WET/DRY
SATURATION
OFF 10
AMOUNT
TONE HRMC.
ENHANCE
OFF
BASS TREBLE
TONE SHIFT
PRE COMP
STEREO
OFF FULL
AIR
MONO SIDES
WIDTH
OFF 500 Hz
MONO BASS
SPAT. COMP SAT.
SECTION ENGAGE
0 dB
-∞ +12 dB
OUTPUT

Softube have long held an enviable reputation when it comes to modelling analogue hardware in software, so there's always plenty of interest when they launch a new product — and that's certainly the case with their latest release. Available in all major native formats and for Softube's Console 1 system, Bus Processor is inspired by the SSL G-series console's famous compressor. And, although its primary role is to deliver that VCA compressor's action and character, Softube have also added some thoughtful extras, including saturation and spatial processing options, making Bus Processor far more versatile than a typical 'bus compressor'.

The main compressor controls are grouped, in a traditional SSL-style layout, in the centre of the GUI, beneath the meter. In a vertical strip to the left of these controls, a wet/dry knob allows for simple parallel compression, and this area also hosts several more controls with which you can manipulate the side-chain signal to refine the compressor's response. The controls for the saturation facility sit within a similar vertical strip on the right, and the bottom strip contains the spatial processing controls, along with individual on/off buttons for each of the main processing sections, as well as a main output level knob.

As well as being fully resizeable, the GUI includes three Extended Controls panels that can be popped open using buttons located in the top-right section of the GUI. These panels provide input and output meters (situated on the left and right of the GUI, respectively) and a further control section at the bottom. Access to the preset system and a global settings page are also available at the top.

Mission: Control

The layout of the main compressor's threshold, make-up gain, attack, release and ratio controls may mimic that of the original SSL, but there are some useful 'modern' twists too. Notably, all of these controls are continuously variable (the original hardware was limited to switched settings), and some of them offer a greater range than on the original. For example, you can set a ratio as low as 1.3:1, which is much more subtle than the original hardware's minimum 2:1. Second, you also get a Knee option and, more interestingly, both Range and Tempo Sync facilities.

Softube Bus Processor

€149

PROS

- The compression section sounds fabulous.
- The saturation and spatial processing are both highly addictive.
- Well-thought-out feature set provides plenty of flexibility and control.

CONS

- None, other than the temptation to overdo the saturation and spatial processing!

SUMMARY

Although, as the name suggests, Bus Processor was conceived primarily for bus processing, it's a powerful 'make it better' tool that could be used on almost any instrument or mix bus.

Engage Tempo Sync, and you can adjust the release knob in musical divisions (beat, bar, or percentages thereof) related to the host DAW's tempo, rather than milliseconds. This has some useful applications, particularly on the drum bus: not only can you link the release time to the dominant beat division of the performance (quarter, eighth or 16th notes, for example) but you can more easily set up the compression to enhance the groove of the drum mix. It's great that you can scale this setting, to fine-tune the action to suit the source.

The Range control allows you to limit the maximum amount of gain reduction that will be applied, regardless of the compressor's other settings. It's not a unique feature but pretty rare on compressors (it's much more common on gates and expanders), and it can guard against things getting too squashed, particularly when Bus Processor is used on the stereo mix bus.

The left-side pane of the main GUI provides a well-specified side-chain processing capability, which can operate on the internal control signal or on an external key input. At the top, you can specify the degree of independence for the left and right channels: set fully clockwise ('linked') both channels react to a summed control signal, whereas at the opposite extreme, the left and right channels respond independently. Next, you can apply a low-cut (high-pass) filter, so that the compressor is increasingly less sensitive to frequencies below the chosen value. There's also a Tone Shift control, which makes the compressor more or less sensitive to the bass or top end. Finally, there's the option to swap the internal control signal for an external signal. So there's considerably more flexibility here than offered by a typical side-chain high-pass filter!

Sound

Of course, the original SSL bus compressor offered just the main compressor control set, and with fewer available settings than here. Its real magic lay in just how well these worked in combination, and the character it imposed on the signal being processed.

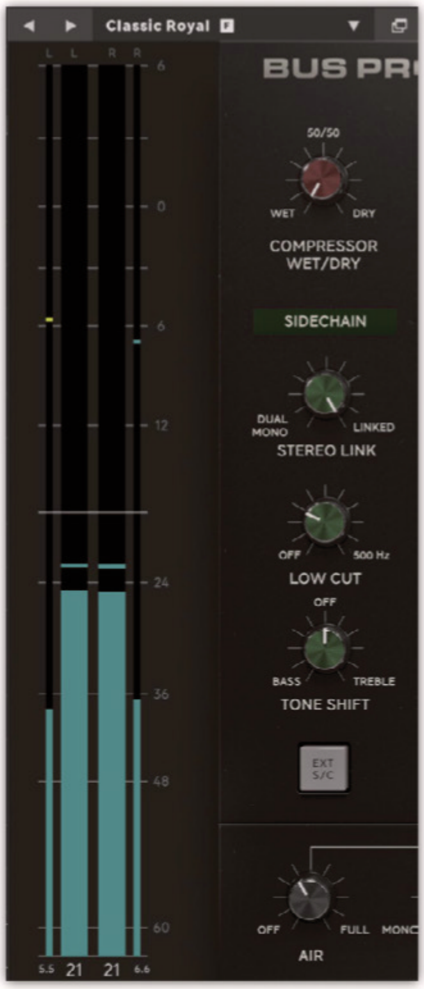
Setting aside, for the moment, the additional control options in Bus Processor's compression section, I'd say that this plug-in is right up there with the very best emulations of this classic circuit that I've used. Indeed, regardless of how faithfully you believe it mimics its hardware inspiration, if you focus in on just this core control set, Bus Processor's compressor is undeniably impressive. It works beautifully on vocals, guitars, piano and bass, for example. It's capable of very transparent results and is very forgiving when pushed a little harder too. However, it is particularly good on a drum bus, where you can dial in all sorts of possible styles and levels of treatment. It also excels on a main stereo output bus, where, with just a dB or three of gain reduction, Bus Processor seems to help almost any mix 'gel' into a more coherent whole. The controls interact in a very intuitive fashion too, making it remarkably easy to arrive at the desired sweet spot.

Used like this, with the other modules bypassed, the Compressor module sounds pretty clean, though of course it does still impart a character in the sense that its dynamics processing may result in tonal changes, as the relative levels of the attack or sustain portions of a sound are adjusted. Naturally, it is possible to dial in a style of compression that's at odds with what your project requires, but it's actually very difficult to 'break' a sound, even when using fairly extreme settings. In short, Bus Processor both sounds and behaves very much like a really good hardware VCA compressor.

Saturation Point

We don't always want 'clean and transparent', of course, and if you are looking to gently adjust the sonic character, Bus Processor's Saturation module has you covered. The saturation can be placed pre- or post- the compression, and is styled on tape distortion. Usefully, you can usually use pretty much the full range of the Amount knob before things start sounding very obviously crushed, although there are some interesting results to be had by pushing the Amount control and then dialling back the saturator's dedicated wet/dry mix.

Softube have a well-established reputation when it comes to modelling the behaviour of non-linear systems and how they distort (do check out their freeware Saturation plug-in if you've not done so already), and the saturation here has a very satisfyingly 'analogue' character. The two Enhance buttons bring slightly different flavours to this saturation, and they can be used individually or in combination. The Tone Shift control adjusts the frequency range within which the saturation is emphasised. Intuitive in use, this affords you considerable control over the contribution the saturation makes to the overall result. The Drive meter, found towards the top of the GUI, lets you visualise just how much saturation you are adding, and can be a very useful reality check. This is undoubtedly one of those addictive, 'more is better' effects that's easy to overdo — as you grow used to it, you'll tend to find that just a little of this pleasing effect can go a long way in adding body or presence to a signal.



The extended panel for the input meters also allows you to adjust the input gain.

Space For More

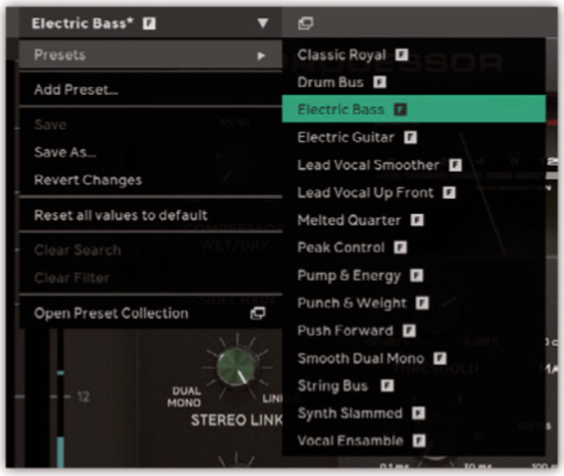
The Spatial module has three controls. Width adjusts the balance between the Mid (fully anticlockwise position) and Sides (fully clockwise) channels of a stereo audio signal, making the whole either narrower or wider. An Air knob boosts the high frequencies, but slightly differently in the Sides and Mid channels,

»

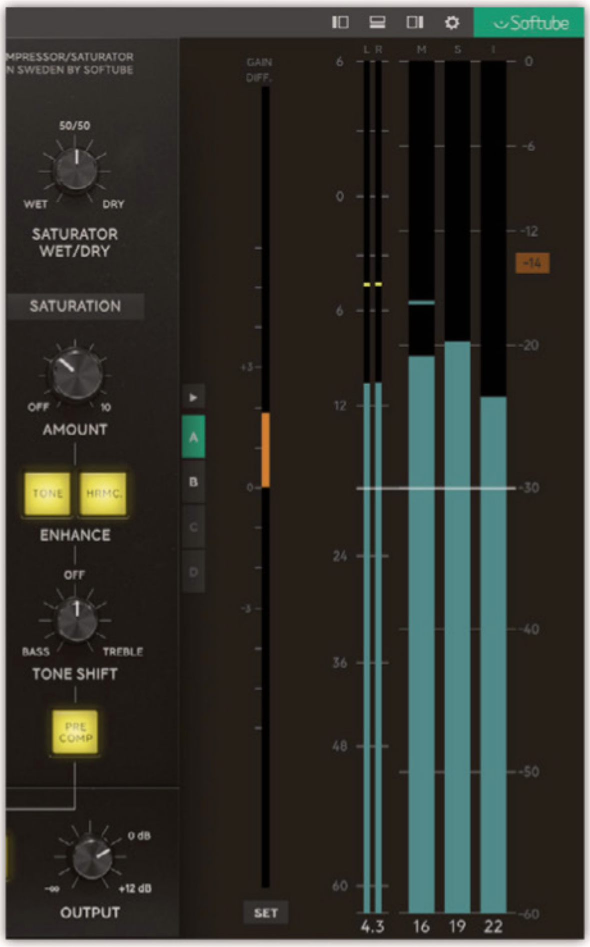
» to add air and sheen, while Mono Bass filters frequencies below the chosen frequency (which can be up to 500Hz) out of the Sides, so bass frequencies remain in the centre of the stereo image and the LF energy is equally distributed amongst the speakers in any playback system.

These are all very simple tools, but they can be incredibly effective, and it makes good sense to include them in a dedicated bus processing plug-in such as this. Applied to an individual instrument bus, you can perhaps afford to be a bit more adventurous with the settings but, as with the Saturation module, a little goes a long way on your stereo mix bus. Still, it's interesting just how much of a 'lift' it's possible to give to a mix by adding just a little Air and setting the Width control in the 110-115 percent range. As a final touch of mix (or master) sweetening to add to that provided by the compressor and saturation modules, this can be very effective indeed, though it's important to understand, yet again, just how addictive this can be, and quickly your ears can grow accustomed to the processed sound: plenty of A/B comparisons are a good way to ensure that you avoid overcooking things.

As mentioned earlier, there are some further controls available on those three pop-out panels, which are accessed using the buttons at the top-right of the GUI. On the left, this gives you input meters along with an input level control. On the right, you get equivalent output metering (you already have an output level control at the bottom of the main GUI). You can configure exactly what both sets of meters display in the Settings menu. For the output meters, that includes options for peak, true peak, RMS and LUFS (Momentary, Short and Integrated, and the options to specify a loudness target). The output meter section also includes a 'gain difference' meter and the Set button. The former shows the perceived difference in level between the dry and processed sounds, while pressing the latter automatically adjusts the output volume to match that of the 'before' signal. This is really helpful, since it allows you fairly easily to assess the sonic changes Bus Processor is making to your signal, without your ears being fooled by any differences in level that result from your processing. The panel also hosts four buttons to store different configurations, allowing you to do comparisons between multiple settings of the plug-in, should you wish.


Electric Bass*
Presets
Add Preset...
Save
Save As...
Revert Changes
Reset all values to default
Clear Search
Clear Filter
Open Preset Collection
Classic Royal
Drum Bus
Electric Bass
Electric Guitar
Lead Vocal Smoother
Lead Vocal Up Front
Melted Quarter
Peak Control
Pump & Energy
Punch & Weight
Push Forward
Smooth Dual Mono
String Bus
Synth Slammed
Vocal Ensamble
DUAL MONO
STEREO LINK

Bus Compressor is supplied with some very useful preset starting points that you can tweak to suit your own audio signals.


Softube
MPRESSOR/SATURATOR
N SWEDEN BY SOFTUBE
50/50
WET
DRY
SATURATOR WET/DRY
SATURATION
OFF
10
AMOUNT
TONE
HRMC.
ENHANCE
OFF
BASS
TREBLE
TONE SHIFT
PRE COMP
0 dB
-∞
+12 dB
OUTPUT
GAIN DIFF.
A
B
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D
SET
L R
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-14
4.3
16
19
22

The output meter panel includes options for loudness metering, as well as the very useful Gain Difference meter and Set button.

Finally, the pop-out bottom panel includes controls for Headroom (this adjusts the internal headroom, and reducing it allows the plug-in to be driven harder for a given input level), High Pass (sets the frequency of a high-pass filter on the plug-in's output) and Phase Invert (allows you to invert the polarity of the sound above the High Pass frequency). My own inclination was to leave this last setting well alone but if you want to explore it, the option is there!

Best In The Box?

Even if you've never experienced a real SSL bus compressor 'in the flesh', you might well have encountered one of the many alternative emulations from the likes of UAD, Waves, Brainworx and, yes, SSL themselves. Having used a number of them myself, I'd say that they all have something to offer. They have plenty in common, though they do have their differences, whether in terms of the detailed character of the compression or the 'added extras' that each manufacturer has included. I have to say that I'd be happy to make Bus Processor my go-to choice for classic SSL-style bus processing.

The compressor itself sounds fabulous. It's also very forgiving and intuitive in use and, as a consequence, it's relatively easy to dial in different styles of compression treatments to suit your needs. The Saturation and Spatial tools are similarly easy to use, very effective and sonically seductive. So it's hats off to Softube's design gurus: this is a really great combination of features delivered with considerable class. It may not be the cheapest SSL compressor emulation you can buy, but it's a brilliant one that more than justifies the asking price.

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Realistic Strings

FAKING A STRING SECTION IN YOUR DAW

MIKE SENIOR

We explain how to create MIDI string arrangements that don’t sound like MIDI string arrangements!

A lot of musicians like using string ensembles in their productions, whether it’s for sweeping orchestral soundtrack cues, anthemic rock/pop choruses, or slippery disco-funk countermelodies. Of course, most of us don’t have a real string section on speed-dial; neither do we have a recording setup comprehensive enough to do it justice even if we did! So we ruefully fire up some sample library or virtual instrument and hope for the best — a tactic that rarely ends very well, to be honest. Indeed, the majority of ‘canned’ string-ensemble parts I hear coming out of project studios only sound at all plausible if you bury them in the mix. The moment you try fading the tracks up as a feature, their manifest fakery just makes your whole record feel cheap.

But it doesn't have to be this way, because there are ways to get remarkably realistic string-ensemble sounds when working with MIDI strings, and in this article I'd like to suggest a few tips that can really help in practice.

Divide & Conquer

The first thing to realise is that you can't just play your strings patch the same way you would a synth patch if you want best results. String ensembles don't play like that, because each instrument section operates independently, tailoring the phrasing and tone to suit the nature of the line, whereas all the keyboard player's fingers are being driven by the same musician!

So one of the things I always recommend is splitting out each internal MIDI line as a separate track. For a start, this gives you much more flexibility to adjust the timbre of the ensemble sound to suit the musical context. You see, the ranges of the different string instruments overlap a great deal, but if you feed your MIDI data into an all-in-one 'ensemble strings' patch it'll give you no choice about which instruments play which notes. There's a world of difference between how Middle C sounds on double basses and how it sounds on violins, and part of the reason that canned strings sound a bit rubbish is that they don't reflect this reality. In particular, melodic parts will sound a lot more convincing if they're not hopping around between different instruments.

And speaking of melodic parts, the other big reason to split each MIDI line onto a separate track is that this makes it possible to take better advantage of the more advanced articulation and phrasing options provided by your choice of virtual instrument. Orchestral libraries now often provide special legato patches, for instance, which render overlapping MIDI notes as slurred phrases, something that can immediately improve the sense of musical realism. Things like accents, staccatos, and swells are also usually much better handled using dedicated articulation-specific patches than by simple MIDI programming tweaks, and it's useful to have the flexibility to switch between those articulations on a per-part basis, rather than just switching the whole ensemble at once.

Velocity Versus Volume

That said, there is plenty you can usually do to improve a string ensemble sound with your MIDI programming too. The most important consideration, in my view, is how you shape volume/expression and velocity data in conjunction with each other. It's vital to realise that volume and expression data typically affect only the level of a string section (much like your DAW's fader automation), but that velocity data will also change the string section's timbre — typically by making higher-velocity notes more strident, as you'd expect if the player were playing with greater intensity.

With this in mind, I usually prefer to adjust velocity data first to achieve a suitable overall timbre and believable musical phrasing, and only then balance the parts against each other (and indeed the rest of the mix) using volume/expression data or DAW fader automation. Mind you, if you're using budget-friendly virtual instruments, you may find that they only have two or three different samples available per note to cater for all 127 velocity values, so I do sometimes feel the need to boost/cut the level of some notes with volume data for phrasing purposes in order to avoid unmusically abrupt tonal switches between velocity levels.

Layering Live Parts

No matter how much of a MIDI ninja you are, or how flashy your strings software is, there is nevertheless still a limit to how realistic programmed strings can sound. Yes, you can put together a pretty convincing chord-per-bar string pad with samples these days, but the moment you start including faster runs of notes or more lyrical melodic lines you'll quickly find things sounding more mechanical. Indeed, one of the simplest tricks to getting decent samples-only strings is just to avoid those kinds of arrangement moves in the first place! When you can't, though, it's time to turn to a stock-in-trade trick that media composers have been using for decades: conceal the fakeness of your string patches by layering a single live player over them — a much easier recording job than tracking a whole ensemble! This is something I've had great success with on a number of occasions, so I have plenty of advice to pass on here as well.

To start with, it's important to direct your efforts efficiently. You don't need to track up live parts for every single line in the arrangement — just focus on those parts that most clearly reveal the shortcomings of your MIDI strings. In practice, the first violin is often a prime candidate for overdubbing, but if that part's just playing some high held note while important countermelodies are happening in the second violin, viola, or cello parts, then a live first violin will hardly help the sense of realism at all. Similarly, there's no rule that says you have to record the whole of the arrangement for a given part — if most of the song just has block chords, then just overdub on the section that doesn't.

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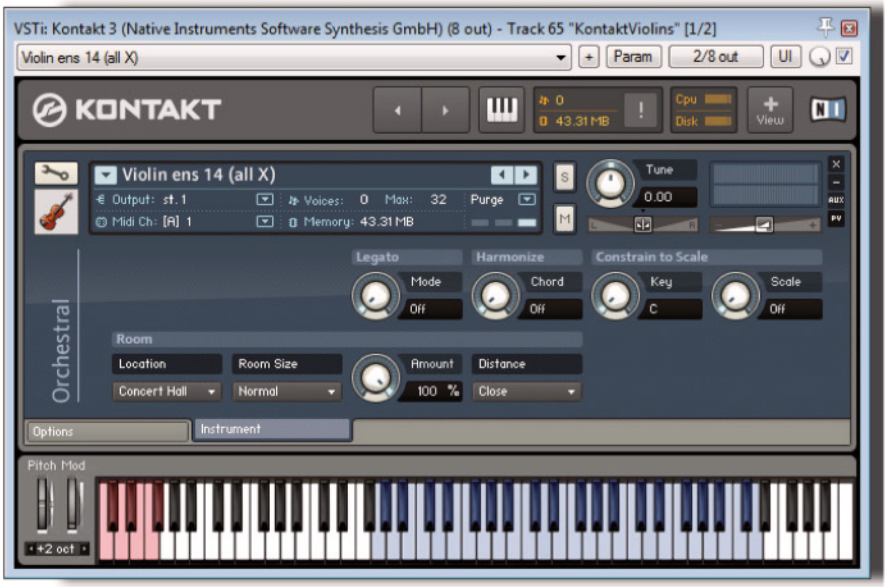

VSTi: Kontakt 3 (Native Instruments Software Synthesis GmbH) (8 out) - Track 65 "KontaktViolins" [1/2]

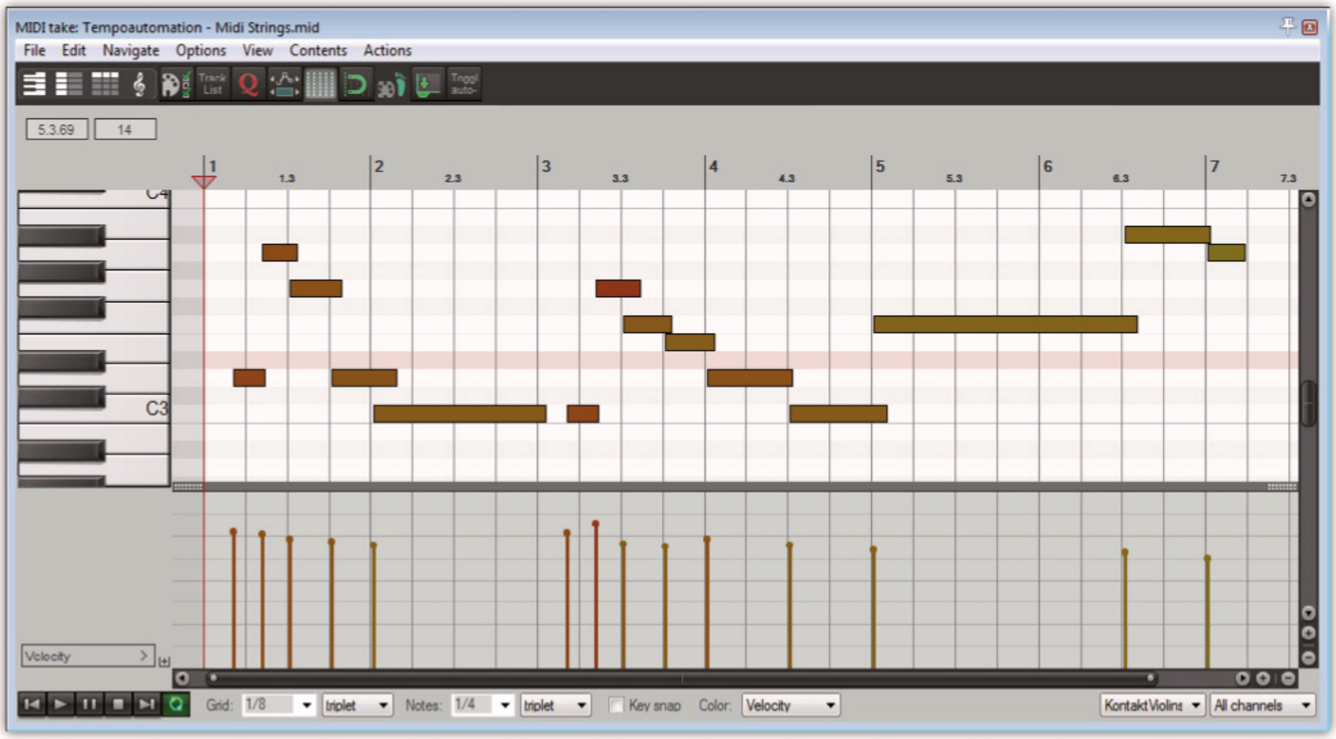
Violin ens 14 (all X)

Output: st. 1
Voices: 0 Max: 32
Midi Ch: [A] 1
Memory: 43.31 MB
Tune 0.00

Legato — Mode: Off
Harmonize — Chord: Off
Constrain to Scale — Key: C; Scale: Off

Room — Location: Concert Hall; Room Size: Normal; Amount: 100 %; Distance: Close

More advanced string-ensemble sample libraries offer keyswitches that can select different articulations. Making use of these can help contribute to a more convincing-sounding string arrangement.



Grid: 1/8 triplet Notes: 1/4 triplet ☐ Key snap Color: Velocity Kontakt Violins All channels

Here you can see how one of the parts in a string-ensemble arrangement has been split out, and then had some of its notes overlapped to take advantage of the more musical note transitions in a virtual instrument's 'legato' mode.

» Now, if you're on a budget, you may well be taking advantage of a friend who used to be a string player at school, but who now only picks up the instrument once in a blue moon — there are a lot of those players around, me included! In this case, it's really important to realise that orchestral string instruments take a while to 'wake up' when they've not been played for a few months. I can only speculate as to exactly why this is, but I imagine that the wood simply becomes a bit stiff when it's not been resonating in a long time. And this isn't me being petty: high-end string-instrument dealers actually employ people to play their instruments regularly so that they remain on top form for potential customers. So your first job before overdubbing is just to make sure the musician has actually played the instrument at a decent volume for a good couple of hours, focusing particularly on the lower strings and any specific notes that feel at all dead-sounding, thin, or reedy. Believe me, if you try to record anything before doing this, your captured sound will definitely suffer.

Mic Technique

For the recording itself, it's important not to mic the instrument too close, otherwise the close-miked sound won't match the more distantly miked sound typical of most string sample libraries. And it's not just a question of reverb levels, because that's not actually such a big issue. The main thing is that the balance of frequencies and instrument facets you pick up a foot away from the instrument is very different from what you get four to six feet back; the more distant placement just sounds a lot more like a real instrument. Now, that said, you should certainly take precautions to keep reverb levels as low as you can (such hanging up a few of those ubiquitous acoustic duvets), especially if you're recording in a small room, simply so that the recording environment's acoustic character doesn't unnaturally colour the captured timbre or conflict unduly with the room ambience of the samples you're using.

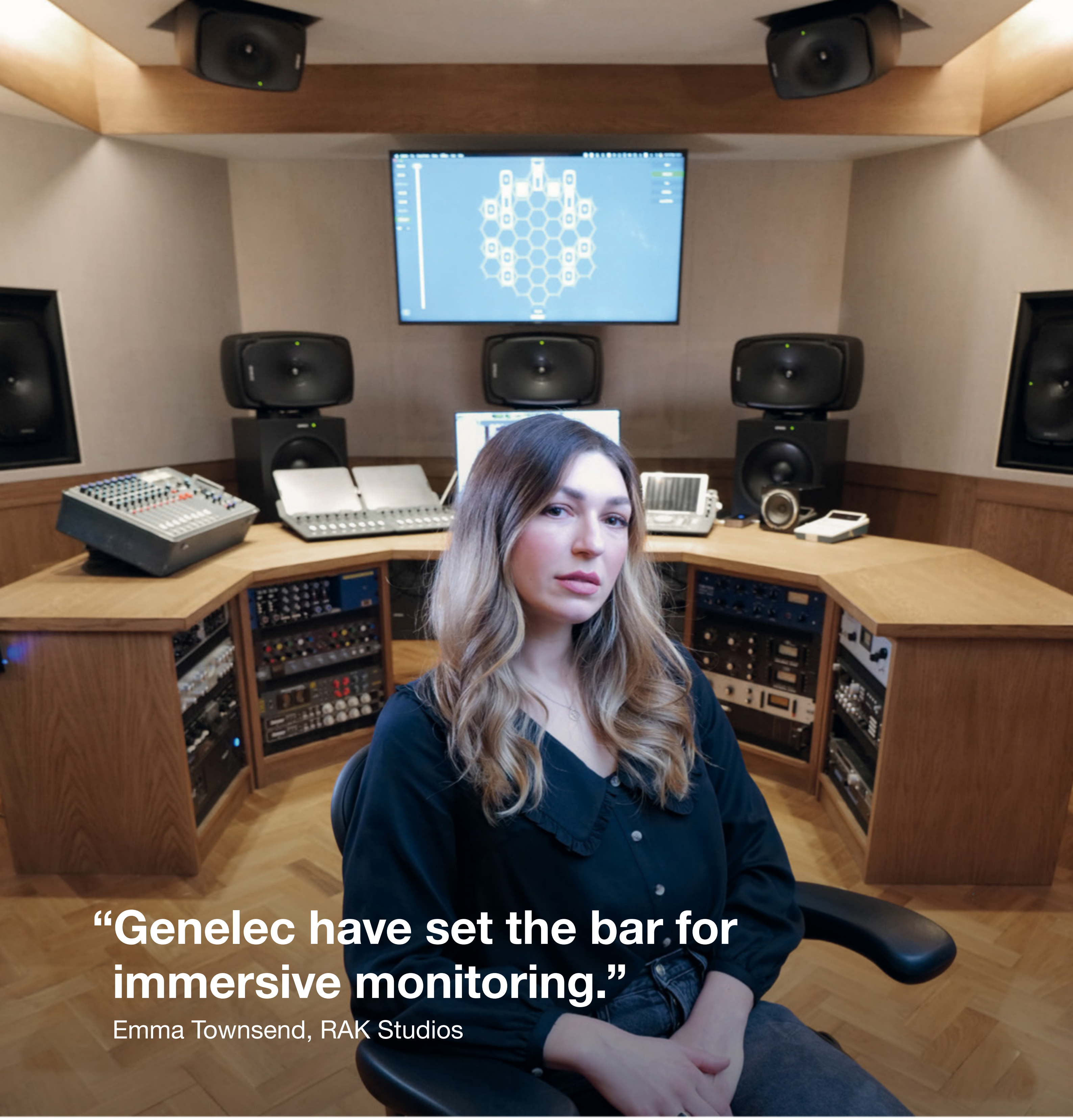
What also helps, in my view, is to record each pass with a spaced pair of mics. Not only does this introduce some image width to the sound (which generally seems to enhance the naturalness of acoustic instruments, for me), but it also stereo-decorrelates any room ambience you do pick up, so that your overdub sounds less like it was recorded in a corridor! Personally, I like to keep the mic spacing quite close, around nine inches, so that there's no risk of mono-incompatibility, and I'm not concerned with trying to generate any kind of 'realistic' stereo image — just a little width and tonal complexity from the two distinct mic positions.

Out of preference, I prefer using omni mics in this application if I can, again because I find that they usually sound more believable and neutral than cardioids. However, I can get away with »

How Low Should You Go?

Within most chart-oriented commercial releases, there's rarely much room to spare at the low end once the kick and bass are in the mix. Furthermore, heavy masking from guitars, keyboards and vocals can often make it seem as though the only part of a string ensemble you're actually hearing is the highest line. As such, it's very tempting to shortcut the job of string arranging by using nothing but violins. I'd advise against that, though, because a big part of the characteristic appeal of strings is the textural complexity contributed by those lower instruments, and that's present even when the listener can't consciously pick out their specific musical lines.

By all means high-pass filter your string parts at mixdown to avoid low-frequency clutter, but keep those violas and cellos in the arrangement nonetheless. Otherwise, your strings will likely sound thin and uninspiring.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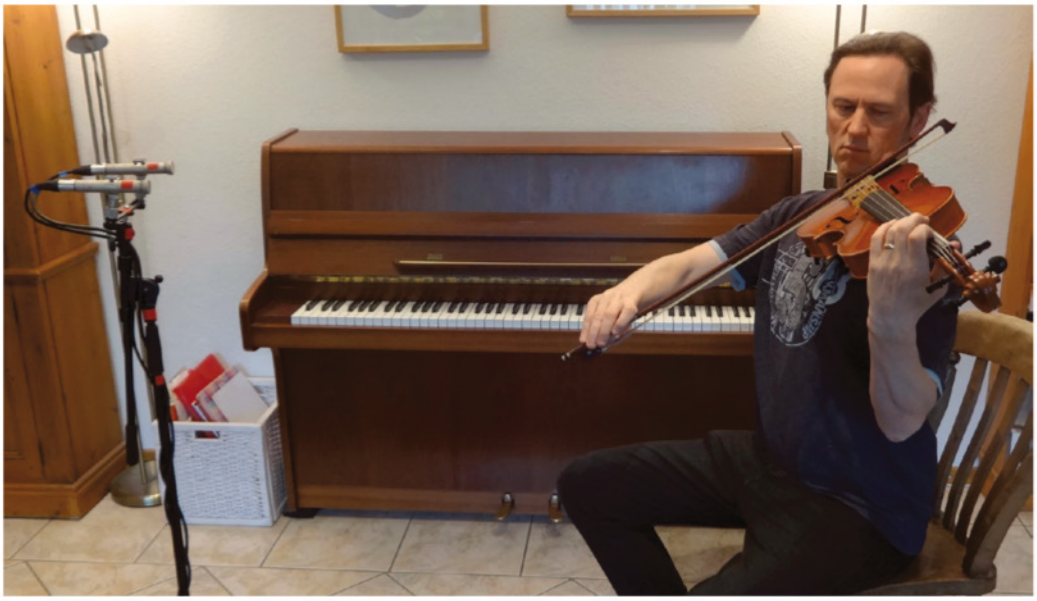

When layering live string overdubs over a MIDI arrangement, recording in stereo from a distance of at least four feet usually helps the recordings fit better with whatever string-ensemble samples you're using.

that because my own home studio is set up as a mix room, so it's acoustically quite dry. If your chosen recording location is more ambient, then directional mics may be the only option. That said, do experiment with figure-eight polar patterns if you have them, as they'll likely pick up the off-axis ambience with less tonal coloration than the cardioids. And, speaking of figure-eight mics, ribbon designs naturally have that polar pattern and are also a great choice for strings, especially with any violin that has a tendency to shrillness — ie. any violin that costs less than your car!

People get very worked up about mic positioning when close-miking string instruments, and with good reason, given how unpredictable the frequency dispersion of wooden-bodied instruments is at close range. However, if you're miking six feet away, the mic positioning tends to be a lot less critical in my experience, and for violin or viola, you can usually set the mics up somewhere in front of the player at chest height and it'll work just fine right away. The same kind of miking position can also deliver good results with cellos, but sometimes moving the mic closer to the floor can help fill out the tone if you want more low-end warmth.

Layering, Editing & Mixing

A crucial question when layering up overdubs of a live string player is how many layers to record per line. Personally, I almost always go for three. If you only do one layer, that single live part will probably feel a bit stranded over the ensemble sound. With two layers, any tuning or timing vagaries between the two parts will sound more obvious, and if you pitch-correct the layers to avoid this, you may then get an artificial phasing-like interaction between them instead. Three layers per line acceptably mitigates all these issues for me. But why not layer things up even more? Well, in my experience, further layers never seem to improve things enough to be worth the extra effort.

Again, if you're working with a string player of the 'weekend warrior' variety, you'll almost certainly want to edit the overdubs to match their takes with the more finessed sound of your samples. Besides, it's much more important to concentrate on the phrasing, tone, and musicality of the live takes, as those aspects of the performance aren't nearly as easily fixable as a bit of dodgy rhythm or pitching. I'm particularly careful to keep my ears peeled to avoid mechanical squeaks from the bow or discontinuities in the timbre of sustained notes, and I also do my best to remind the string player to take advantage of idiomatic performance techniques wherever possible — things like vibrato, bow-slurring, left-hand position shifts, and volume swells.

Once the overdubs are recorded and edited, the lion's share of the job's done, and the mixing process is usually pretty straightforward. Clearly, if there are any obvious tonal disparities between the overdubs and the samples, then some EQ may be called for, and I sometimes find that gently low-pass filtering the overdubs at, say, 15kHz can help sit them a step further back in the mix if they're feeling too dislocated from the rest of the virtual ensemble. (That said, you often need to brighten string ensembles overall within a pop/rock mix context to combat high-frequency masking from things like cymbals, percussion and electric guitars.)

Compression can come in handy, as string instruments can be surprisingly dynamic when recorded comparatively close-up. Even so, you'll almost always need to supplement that with some fader automation at mixdown to refine the balance within the context of the full production.

Beyond that, a healthy dose of artificial room ambience added to the live overdubs will often help blend them better with the samples, especially if your recording room wasn't very reflective sonically. After all, in most real-world situations where you'd hear a string ensemble, a good deal of what reaches your ears will be room reflections rather than direct sound. Some hall reverb applied to the entire string ensemble usually provides a nice final touch, further gluing the layers together into a smooth and coherent sound.

Even Better Than The Real Thing

The suggestions I've presented in this article aren't rocket science (and are a heck of a lot cheaper and easier than trying to record a real string orchestra), but I can tell you from real-world experience that they really do work in practice! To demonstrate this, I've uploaded some audio examples to the *SOS* website at https://sosm.ag/realistic-strings that illustrate the kind of improvements you can be achieved. Plus, if you head over to www.cambridge-mt.com/sosstrings you can download a supporting DAW project where you can experiment with the balance of the samples, live overdubs, reverbs, and backing stems in my demonstration arrangement for yourself. ■■■

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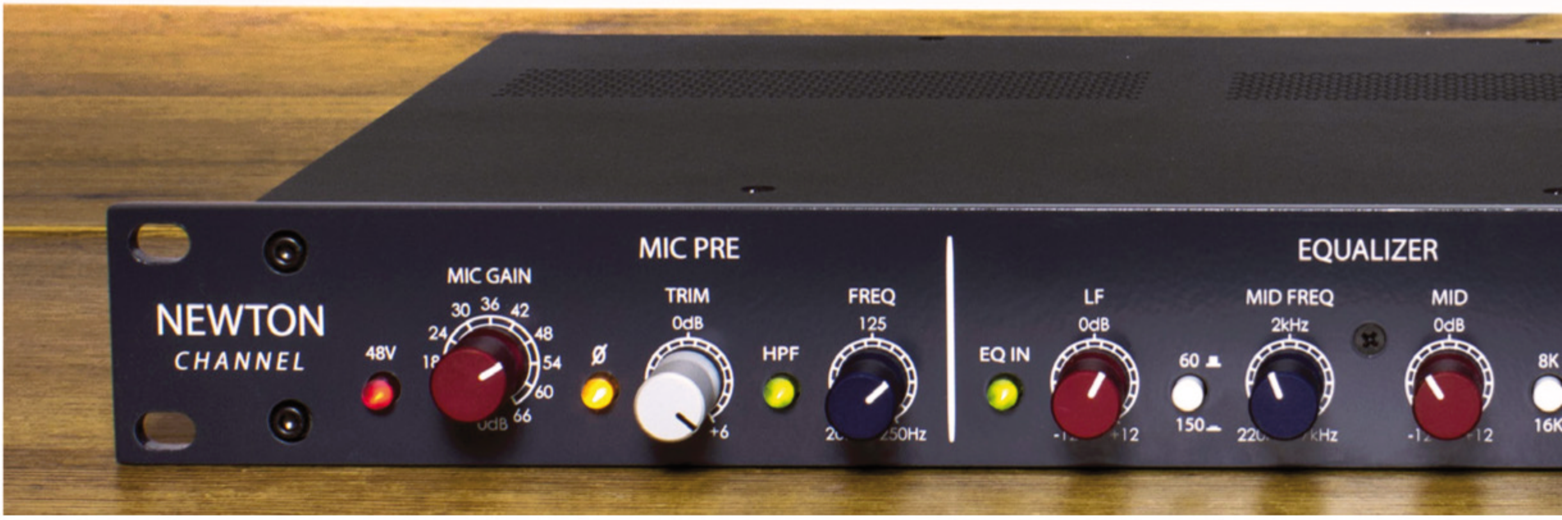
Rupert Neve Designs



BOB THOMAS

Analogue Channel Strip

Though based on Mr Neve's historic designs, the Newton offers something a little different — for a little less money!

When the definitive history of the audio recording industry is finally written, the name of the late Rupert Neve (1926-2021) will feature very prominently. At the present time, Mr Neve's stellar reputation rests primarily on the products he developed during his 10 years at the head of Neve Electronics, the company that he founded in 1965. But much as I love and fondly recall the Neve consoles I've worked on over the years, I'd argue that the crowning achievement of his long career is the Rupert Neve Designs Shelford series. Taken together with the company's 5088 mixing console and all the other products developed under Mr Neve's stewardship of Rupert Neve Designs (RND), this forms a legacy the like of which we will probably never see again.

Fortunately, for musicians and audio engineers alike, Mr Neve's philosophies, techniques and methodologies live on in the engineering and design team, who he trained and mentored and are now, in the words of his obituary on the RND website, the "caretakers of the theories, practices and ideologies that truly constitute a Rupert Neve Design". This rich heritage is exemplified in the new Newton Channel, a combination of mic preamp, three-band EQ and compressor whose design and published specifications exhibit all the characteristics of a Rupert Neve Designs product. Deriving its name from Newton Abbot, the town of Rupert Neve's birth, the Newton Channel may not be inexpensive in absolute terms — quality like this costs — but it is RND's most accessibly priced rackmount channel strip to date. It was conceived for use in project studios, commercial studios and live sound applications, as well as in the home setups of those podcasters, voiceover artists and musicians who wish to invest in an easy to use but high-quality recording front end.

Outside & In

With its dark grey fascia, red, black and silver control knobs, illuminated buttons and white legends, the Newton Channel apes the cosmetic appearance of the Shelford Channel, and would not look out of place sitting in a rack next to its more expensive stablemate. The mic/line preamp section's layout is similar to that the RND 511 500-series mic preamp.

A 12-position rotary switch selects the amount of microphone gain in 6dB steps between 0dB and +66dB, an arrangement that avoids the 'gain bunching' at higher gain levels often exhibited by cheaper pot-based designs. A 31-position detented potentiometer then allows the selected gain level to be trimmed by ±6dB. This results in a total possible gain of +72dB, but also the ability to match very precisely the signal levels across two units for stereo operation. The 48V phantom power, polarity reverse and continuously variable Sallen-Key (20Hz-250Hz, 12dB/octave) high-pass filter are, like other front-panel relay-switched functions, activated by LED-lit buttons.

Unlike other current RND equalisers, the Newton's three-band EQ contains no inductors, featuring instead an updated version of the state-variable filter topology found in legacy Portico series products such as the 5032 and 5033. It can be switched in/out of circuit, and 31-position detented potentiometers provide ±12dB of

Rupert Neve Designs
Newton Channel
£1979

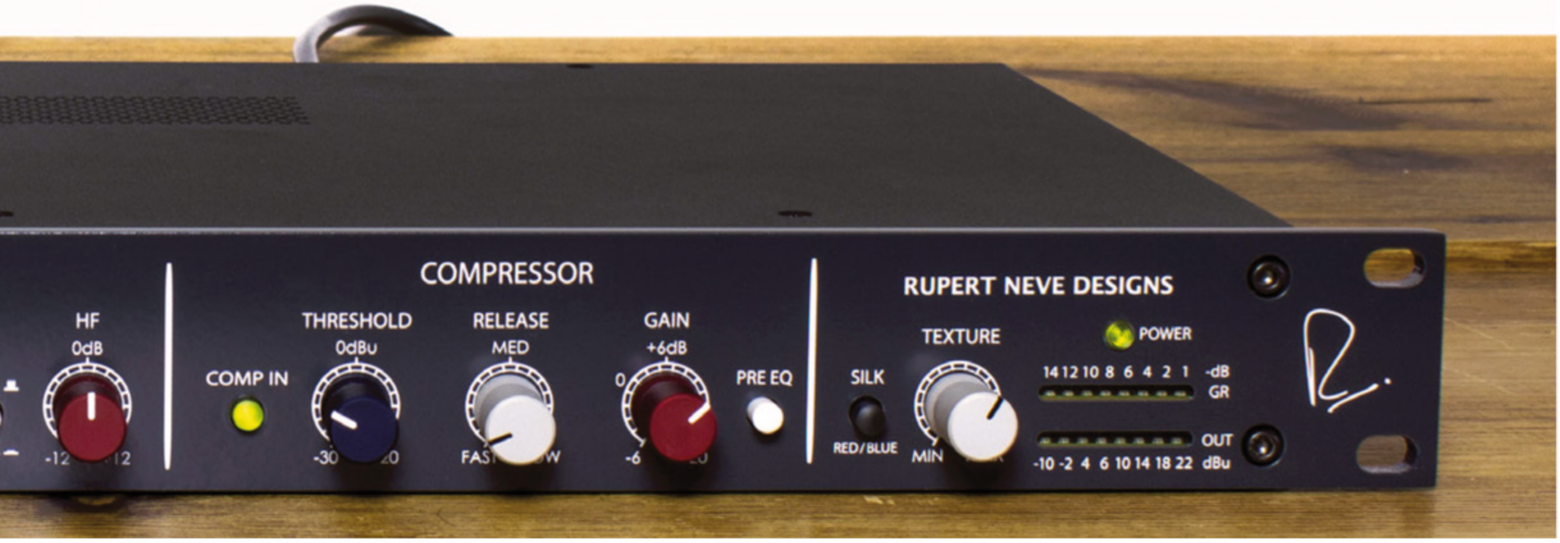
PROS
- Sounds as good as the specs suggest.
- Intuitive controls.
- Output transformer and Silk function sound seductive.
- Good price:performance ratio.

CONS
- None.

SUMMARY

An interesting channel strip that delivers ease of use and impressive performance at a lower price point than might be anticipated.

Newton Channel



gain in each band. The corner frequency of the low-frequency shelf can be switched between 60 and 150 Hz, and that of the high-frequency shelf between 8 and 16 kHz. In between these shelves is the mid band, which has a peaking response and can be swept at a constant Q between 220Hz and 7kHz using yet another 31-position detented pot.

The VCA compressor can be placed pre or post the equaliser. Its modest control setup consists of a trio of (you've guessed it) 31-position detented pots, the first of which sets the compressor's RMS detector threshold (+20dBu to -30dBu). The compression is of the soft-knee variety, with has a fixed ratio of 2:1, and, interestingly, a fixed 20ms attack time; more on that later. The second potentiometer selects the release time, within a range of Fast (50ms) to Slow (500ms), and the third controls the make-up gain (-6 to +20 dB). A rear-panel quarter-inch TRS jack socket allows the compressors of two Newton Channels to be linked for stereo operation.

The final front-panel control section carries RND's signature Silk Texture control, again on a 31-position detented potentiometer. This is scaled from Min to Max and enables you to adjust the contribution of the button-selectable Red Silk or Blue Silk circuits, which add second- and third-harmonic transformer-based distortion to the signal leaving the Newton Channel. Sitting next to this section are two horizontally mounted eight-segment LED displays, the upper of which displays gain reduction (0 to 14 dB) and the lower the output signal level (-10dBu to +22dBu).

The rear panel carries a balanced XLR/TRS combi input connector to the mic preamp, the aforementioned TRS link jack, two transformer-balanced XLR output connectors (one of which puts out a 6dB lower level than the other), a ground-lift switch and a switched IEC mains socket.

Internally, the Newton Channel is a thing of beauty, with high-quality components and a superb PCB layout. Components are through-hole types where that's sonically important, and surface-mounted where not. As in several other RND products, the Newton is fitted with a pair of high-quality, Japanese-manufactured, switch-mode power supplies, configured to create the necessary bipolar power-supply voltages. As with other RND devices, the Class A amplification stages are used throughout the signal path.

The custom-designed RND-2042 output transformer features twin taps, one of which supplies the main output (max +23.6dBu at 1kHz) whilst the other delivers the lower -6dB output (max +17.6dBu at 1kHz). Although these numbers equate, more or less, to standard operating levels in the US and Europe respectively, the main purpose of the dual outputs is not to simplify the lining up of hybrid analogue/digital signal chains on either side of the Atlantic! Rather, the lower-level output allows you to 'push' the Newton's Class A amplification stages and output transformer harder, in order to obtain the much-loved second- and third-harmonic distortion that such actions (in concert with the Texture/Silk functionality) can deliver, without your enthusiasm leading you to clip the input stage of the next unit in your recording chain.

In Use

As I noted above, the Newton Channel is RND's most accessible channel strip in terms of its price, and it is arguably also their most accessible and intuitive in terms of operation. Although I record in the digital domain, I tend to use a lot of analogue

»

Technical Performance

The specifications in the Newton Channel's rather impressive listing are accompanied by the conditions under which they were measured, a trait that I wish that all manufacturers of audio equipment would follow, as it makes comparisons between competing products both easier, and more meaningful. Given that these measurements are freely available on the RND website I won't repeat them all here, but highlights include (input to output, unweighted and without EQ, compressor or Silk in circuit) an EIN of -125dBu, a frequency response that is within -3dB from below 5Hz to 140kHz and, more relevant in terms of human hearing, within ±0.1dB from 10Hz to 30kHz; and a THD+N (<10Hz to 22kHz) of 0.0013% (1kHz, 0dBu).

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GND
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MAIN OUT
LINK
INPUT

The Newton Channel has two outputs, one being 6dB lower in level than the other, to make it easier to manage the level added when driving the Silk and transformer stages harder to add character.

hardware in preference to plug-ins — so I spend a lot of time looking at VU meters. I run the digital side of my studio at the European standard of +18dBu = 0dBFS and line up at +4dBu = 0VU, which leaves me 14dB of headroom below 0dBFS. With the Newton, lining up at that level leaves me with limited meter resolution of signals below +4dBu. However, if I switch over to the secondary -6dB output I can line up the Newton so that +10dBu = 0VU, thereby increasing the meter resolution available for lower level signals so it's on a par with that available for higher level signals. This still leaves me approximately 14dB of headroom within the Newton before hitting 0dBFS.

To my ears, the transformerless mic preamp delivers a clean, neutral and inherently modern-sounding performance. Its high input impedance (8.9kΩ) minimises the load on the microphone, and helps to preserve transients and deliver a flatter frequency response. The result is that the Newton places the sonic character of the connected microphone very firmly front and centre. Actually, I should say 'connected source' because, with 72dB of gain available and an ability to handle input levels of up to +23.6dBu, both microphones and line-level devices can be used with this preamp. Inevitably, given that the Newton's low-frequency response extends to below 5Hz, the high-pass filter will routinely be used to keep handling noise and other unwanted low-frequency sounds out of the sonic picture. One final point worth noting about the preamp is that the 48V phantom power supply can, depending on the connected microphone, take a few seconds to come online after switch-on.

With no inductors anywhere to be seen, the three-band EQ also displays a definite air of modernity. There's ±12dB gain for each band, two available corner frequencies (an octave apart) for both the treble and bass shelving bands, and a five-octave peak-response midrange sweep. This all combines to create a musical-sounding EQ that I found to be easy on the ear and intuitive in use. Changing the switched corner frequency combinations changes the areas where the shelving filters overlap with the peak response midrange, of course, creating four subtly different equalisation profiles that add to this equaliser's flexibility. Once I became familiar with this, I found that I could achieve the results I wanted quite quickly and intuitively. The ability to switch the EQ in and out to check that I was enhancing rather than destroying a source is another extremely useful feature.

With its fixed 20ms attack time and 2:1 ratio, the Newton's feedback VCA compressor may theoretically be somewhat limited in terms of the facilities offered, but the practical reality is that this is an easy-to-use compressor that allows fast transients to pass through unsullied, while delivering natural and smooth-sounding dynamic control. Operation quickly becomes intuitive: you can think of it as being much like a 'one knob' compressor, but with the added benefit of a variable release. Once you have a feel for how the controls react, it's very easy to dial in the amount of gain reduction you wish to hear: adjust the release time to suit the source, match the compressed and uncompressed signal levels, and switch the compressor in and out to compare the result with the uncompressed original.

Until you've actually used Rupert Neve Designs' Silk function, it's easy to underestimate just how effective and addictive it can be! With a development history that stretches back to the Warmth, Glow and Sheen controls that Mr Neve developed during his time working for console manufacturers Amek, Silk is an ingenious means of adding character to a signal. They've asked that I don't give the finer details away but, essentially, it allows for flexible manipulation and filtering of the custom transformer's pleasing harmonic content. The result is an emulation of the sonic character of Mr Neve's early Class A, transformer-coupled designs, and it allows you to add either warmth and weight in the low/low-mid frequencies, or a sheen and sparkle in the higher frequencies — it's all achieved simply by turning the Texture control and switching between the Red and Blue Silk modes.

Summing Up

The Newton Channel is an interesting move for Rupert Neve Designs. Its overall sonic performance is impressive and it displays all the quality and characteristics you'd expect of an RND design. It also allows you, quickly and intuitively, to achieve great-sounding results. This, together with the Texture/Silk combination, distinguishes the Newton Channel from its obvious competitors and, to my way of thinking, more than justifies its price. It may not be what most readers would call inexpensive but it is, as I mentioned earlier, the most affordable complete channel strip RND have produced. So if you have the budget available and are in the market for a high-quality, versatile, mic/line channel strip that is easy to use and can add warmth, colour, sheen and sparkle to taste, I would strongly recommend that you take a closer look at the Newton Channel.

ALTERNATIVES

To my mind, the Newton Channel's closest, but not directly comparable, competitors are the slightly more expensive **Manley Core Reference** and **Gainlab Bishop** channel strips, and the slightly more affordable **Tegeler Audio VTRC**.

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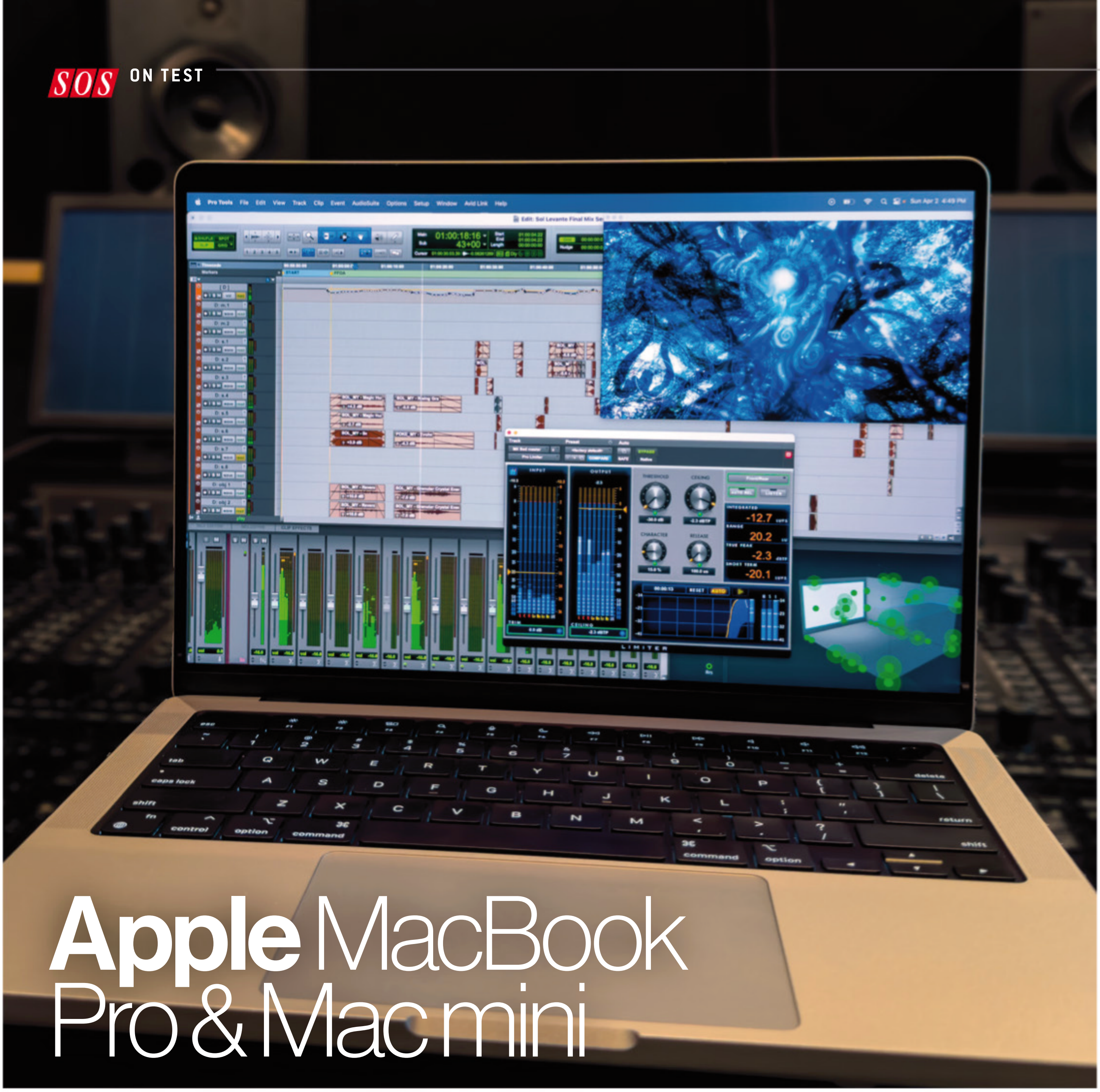


Apple MacBook Pro & Mac mini

Personal Computers: Part 2

We put the latest M2-based Macs through their paces with music benchmarks.

MARK WHERRY

In last month's *SOS*, we looked at the new M2 Max and Pro chips that are powering Apple's latest generation of computers, and ran some general performance benchmarks on the latest M2 MacBook Pro and Mac mini models. But how do those machines perform with the music software that we use every day? We begin our testing with Logic Pro, which last year celebrated its 20th anniversary under Apple's ownership, and which was, unsurprisingly, the first popular music and audio application to support Apple Silicon.

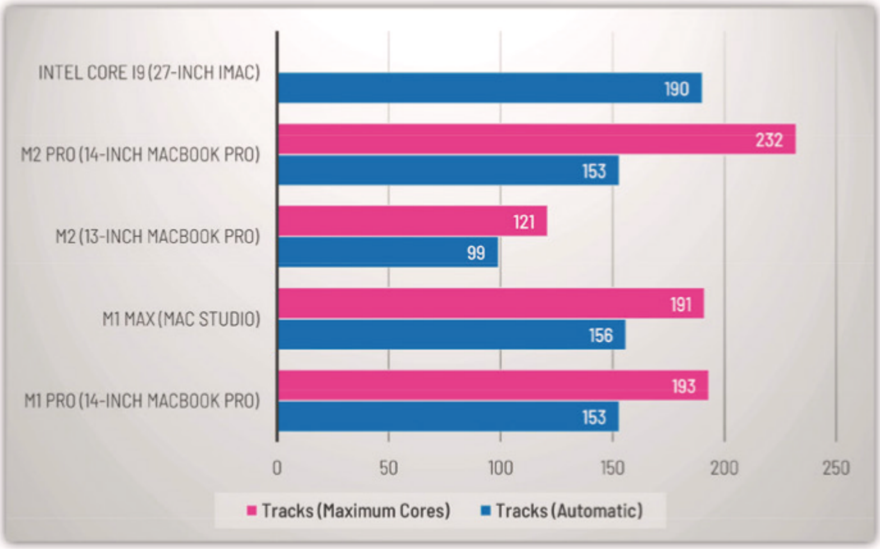
Logic Pro 10.6 shipped as a Universal Binary — a single application file containing versions for both Intel and Apple Silicon-based architectures — and can seamlessly run Audio Units out of process that are written for Intel systems, which have yet to make the transition, whilst Logic Pro itself executes the latest Apple Silicon code. This approach is similar conceptually to the so-called 'bridges' that were common during the 32- to 64-bit transition over a decade ago, although Logic Pro's implementation is transparent to the point where you probably wouldn't initially realise what is actually happening.

Real-world tests in a music-making environment are often more revealing than general benchmarks!

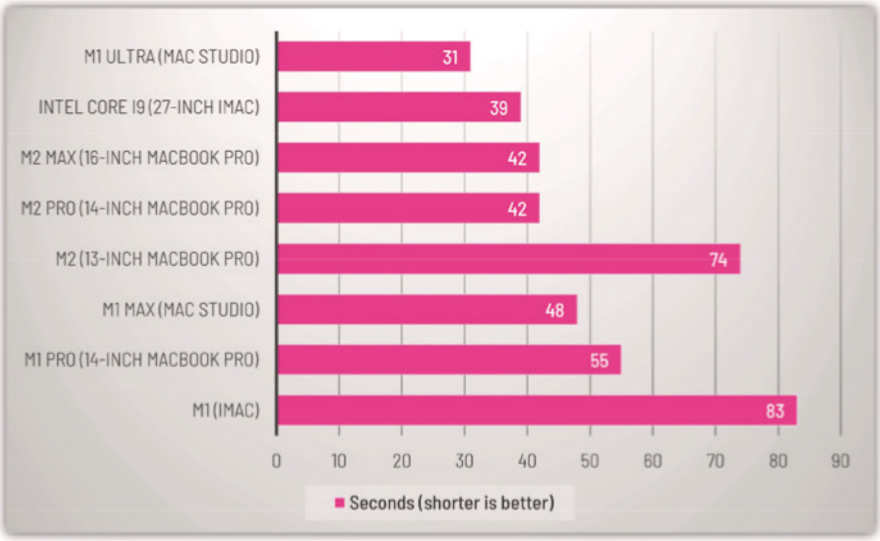
Apple sometimes quote Logic Pro performance when comparing the abilities of newer Macs to earlier models, most recently for the late-2021 introduction of the 14- and 16-inch MacBook Pros, which were tested with a pre-release version of Logic Pro 10.7.0, running a Project "consisting of multiple tracks, each with an Amp Designer plug-in instance applied. Individual tracks were added during playback until CPU became overloaded."

Sadly, this information hasn't been updated for the new M2 Pro and M2 Max-based models, so instead, I've used the tests described by Apple as the basis for my own, using the Amp Designer plug-in as an insert and choosing the Britannica Crunch preset, as befitting myself.

Apple don't specify whether they used mono or stereo tracks, so I decided to save myself some time by using stereo tracks, playing back a 44.1kHz/24-bit audio file. In the interest of limiting the number of tracks to a somewhat sensible number, I used two instances of Amp Designer per track. Using one insert instance per track would have placed a greater burden on the way the audio engine schedules its processing across the available cores, so this was my attempt to balance the performance between scheduling and processing. It's probably also more representative of real-world usage, since it's more common than not for tracks to have more than just a single insert.



Here are the stereo Logic Pro track counts (each with two instances of Amp Designer) for various Macs. It's interesting to note that the Intel Mac still offers respectable performance (although accompanied by industrial-level fan noise), while the new M2 Pro really shines. Note that the Automatic setting for Intel-based Macs seems to use the maximum number of cores, which makes sense given their heterogeneous performance.



This chart represents the time required to perform an offline bounce (in seconds) of the Montero Spatial demo song included with Logic Pro 10.7, leaving the Audio Device Processing Threads option set to the default Automatic configuration. Note that the M1 Ultra handily bests its siblings, although the last-generation Intel iMac (based on a Core i9 processor) earns an honourable mention in second place.

Of Threads & Cores

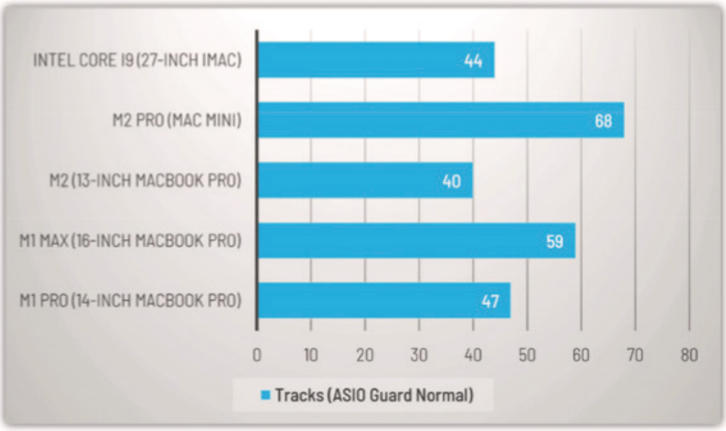
Logic provides the user some control over how the application's audio engine uses the available processing power, courtesy of the Processing Threads setting in the Audio Devices Preferences window. For Intel-based Macs, which used processors with symmetric cores that all offered the same level of performance, this setting was relatively straightforward. If you had a 10-core processor, you could set Processing Threads to '10' and the audio engine would scale its workload evenly across the processor's 10 CPU cores. And while this scaling wasn't linear, such that setting the number of threads from five to 10 would double the performance, for example, it would allow Logic Pro's audio engine to push the machine to the absolute maximum whilst juggling all the other threads required by the system to run the application, operating system, and everything else.

With Apple Silicon Macs, the Processing Threads setting behaves slightly differently, due to the asymmetric nature of the CPU cores designed for efficiency and performance. By default, Processing Threads is set to Automatic, meaning that Logic will choose the best option for balanced performance based on your Mac's CPU resources. And while you can still set the number of Processing Threads, that number now determines whether the efficiency cores get used as well as the performance cores.

For example, on a 12-core M2 Pro-based system, with its eight performance and four efficiency cores, setting the number of Processing Threads between two and eight means only the performance cores get used, whereas setting this to 10 or 12 allocates either two or four efficiency cores as well. On the Mac mini, for example, I was able to play back 153 tracks with Processing Threads set to Automatic. Setting Processing Threads to use the maximum number of cores increased the performance of the audio engine substantially, making it possible to play back 232 tracks — a 50 percent improvement!

It's tempting to conclude that the Processing Threads setting essentially behaves the same on Apple Silicon-based Macs as it did on those with an Intel architecture, such that more cores equals more audio processing power, but such an outcome isn't always the case. For example, if I reduced Processing Threads to use 10 cores (employing the same number of performance cores, but with two fewer efficiency cores) I got the same result of 232 tracks. And, based on the plug-ins you're using in your own Projects, you may find situations where the performance improves if you don't include the efficiency threads in the processing pool.

With Processing Threads set to eight, such that only the eight performance cores are used, I was able to play back 212 tracks simultaneously; reducing the number of threads to six, four and two, it was possible to play back 168, 103 and 69 tracks



With ASIO Guard set to Normal, the M2 Pro shows off how well Cubase Pro's audio engine balances the utilisation of performance and efficiency cores.

A Level Playing Field

To evaluate the performance of the new Mac mini and MacBook Pro models, I had similarly specified M2 Pro-based models available for testing (each 12-core systems with 32GB memory, see 'Test Spec' box). In order to make comparisons with previously released systems, I also used a base model 13-inch MacBook Pro (equipped with an M2), a 14-inch MacBook Pro with an M1 Pro (the previous base model for this configuration), a 16-inch MacBook Pro and Mac Studio (with the same M1 Max configuration featuring 64GB memory), and a 27-inch iMac (the Intel-based iMac with a 10-core i9 processor, 64GB memory). All systems were running Mac OS Ventura (version 13.2.1).

Each Mac was tested using its own built-in hardware via Core Audio, clocked internally at 44.1kHz with a buffer size of 256 samples. And default settings were used in each application unless otherwise specified.

Finally, in addition to the usual real-time performance tests, I also decided to include some offline tests. Because while being able to run a large number of effects and instruments is obviously vital to modern production, being able to quickly perform an offline bounce is also valuable and should be a great way to test peak performance. In an ideal world, where real-time processing only needs to be able to perform enough computation within a given window of time — as dictated by the size of the audio buffer — the perfect offline process should, at least in theory, reveal something about the absolute peak performance of a system given that it can process as much data as it wants as fast as possible.

simultaneously. Therefore, based on these results, I'd be inclined to restrict Logic's audio engine to use only the performance cores as my default setting, and only experiment with adding the efficiency cores into the mix once the dreaded "System Overload." alert appears — an alert so foreboding that it even includes a full stop in its title.

On The Bounce

Moving along to offline performance, I used the Montero Spatial Project, which can be downloaded and opened from within the Demo Projects category in the New Project Template window. On the M2 Pro-based Mac mini, the demo Project in question (which is 2:51, or 171 seconds in length) took 42 seconds to bounce. This is roughly twice as fast as performing the same operation on an iMac with an M1 processor, which took 83 seconds, and even marginally quicker than the 48 seconds it takes using either a previous-generation 16-inch MacBook Pro or current Mac Studio with an M1 Max chip.

However, it turns out that this isn't the quickest time achievable when bouncing, since Logic's offline processing uses the same configuration that's set for real-time processing in the Audio Devices Preferences window. This means that if the Processing Threads setting is set to use the lowest number of CPU cores, bouncing will take longer than if this setting is configured to use a higher thread count.

Test Spec

Apple Mac mini with M2 Pro (12 CPU cores, 19 GPU cores), 32GB unified memory, 1TB SSD, and 10G Ethernet.
Apple 14-inch MacBook Pro with M2 Pro (12 CPU cores, 19 GPU cores), 32GB unified memory, and 2TB SSD.
Tested with Mac OS Ventura (13.2.1) and Apple Studio Display.

So, on the same Mac mini, the same bounce took 114 seconds with Processing Threads set to two (a 170 percent increase), whereas I could achieve the bounce in 40 seconds if Processing Threads was set to use the maximum 12 cores.

This is worth remembering, since with Intel-based Macs you'd probably leave the Process Threads setting left to the default, maximum number of CPU cores, whereas on an Apple Silicon Mac, the default, Automatic setting won't give you the peak performance possible for offline rendering straight out of the box.

Steinberg's Cunning Plan

Steinberg introduced native support for Apple Silicon in Cubase 12, the current release version being 12.0.52 at the time of writing, although the Intel version of the application can of course still be used under Rosetta 2 should the need arise. One reason this might be useful is that, in the company's quest to move past the VST 2 plug-in technology they first introduced in 1999, Cubase 12 only supports VST 3 plug-ins on Apple Silicon.

Although Cubase is a cross-platform application, version 12 fully embraces many key Apple technologies to provide Mac users a first-class experience, and perhaps the best example of this is in Steinberg's support for Apple's Metal hardware-accelerated graphics framework for drawing the user interface, which was first introduced in Cubase 11. This enables the application to take full advantage of the GPU cores in presenting a fluid and responsive visual front end without having to burden the CPU cores, and it makes a significant impact on performance, as can you see in the iStatistica Pro screenshots. When running with Metal GPU-based acceleration, Cubase required 58 percent less CPU resources for graphics processing at a given point in the playback of the Pop Demo Project included with version 12.

Moving on to audio performance, I recreated a similar test project to the one used for Logic Pro; however, since Cubase obviously doesn't include Logic's Pro's Amp Designer, I turned to a similar plug-in Steinberg include called VST Amp Rack, selecting the Big Clean Reverb preset. The reason for sticking with guitar-amp simulations is because they generally employ convolution-based filtering techniques that, if you'll pardon the pun, amply put the CPU through its number-crunching paces. Given the VST Amp Rack includes virtual effects pedals, which Logic Pro includes separately in its Pedalboard plug-in and Stompbox collection, it therefore has a significantly higher CPU overhead than Amp Designer, although I also used two instances per track so as not to front-load the scheduling carried out by the audio engine.

Keep Your Guard Up

Unlike Logic Pro's Processing Threads setting, Cubase doesn't provide the user with an option to configure or

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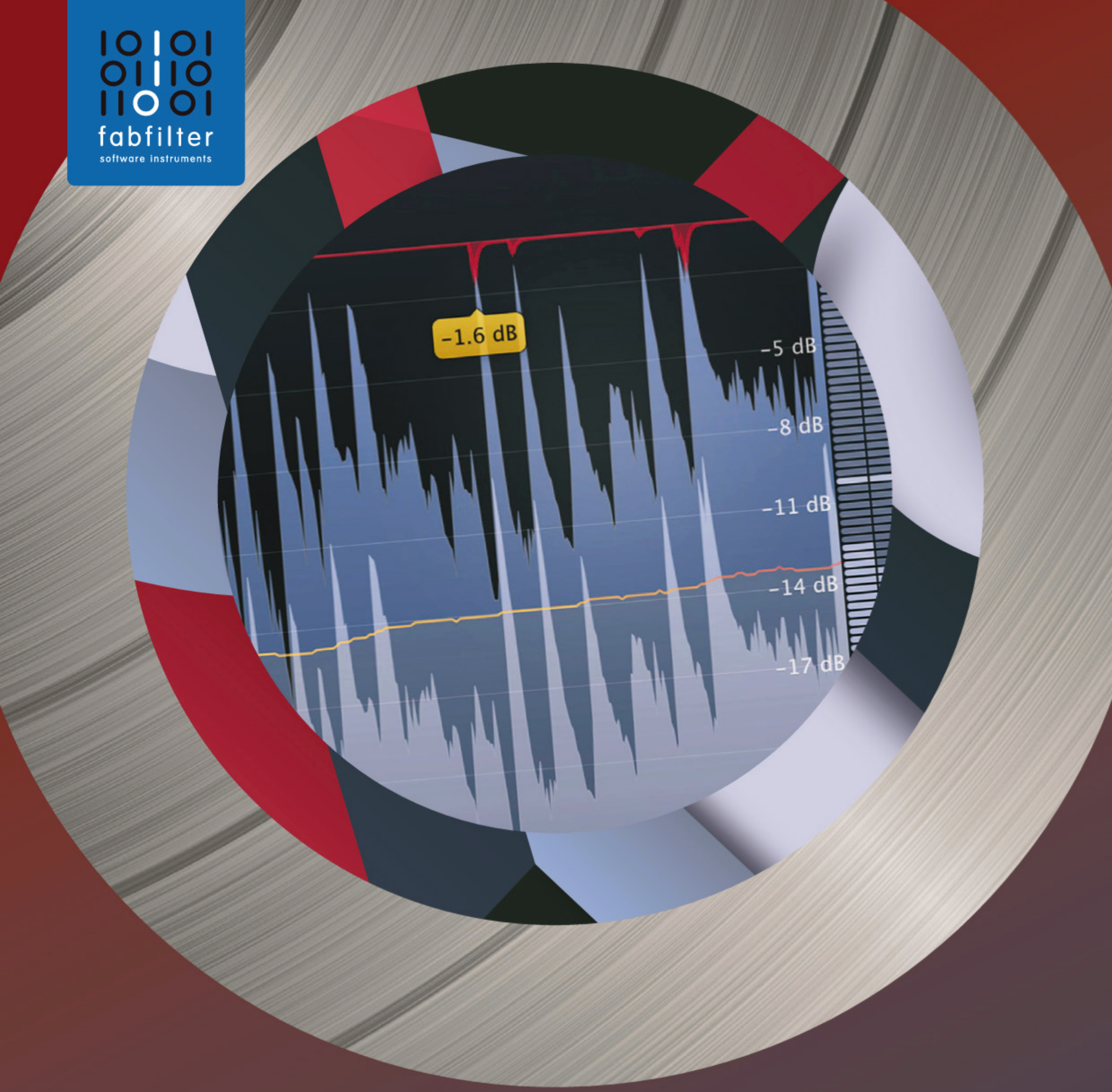
software instruments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» constrain how many performance cores are employed, or to specify a balance between performance and efficiency clusters. However, there's a good reason for not offering such a control, largely informed by a technology Steinberg brand as ASIO Guard.

ASIO Guard implements a 'prefetch' system, which, put simply, is a bit like your dog returning a stick to you before you've even thrown it. In the case of Cubase's audio engine, ASIO Guard enables the system to look ahead from the current playback position and deploy any free processing resources to speculatively pre-render audio as it might be needed.

With ASIO Guard disabled, the Mac mini could play 43 tracks. Cubase's Audio Performance window, which has been overhauled for Cubase 12 to give the user more meaningful feedback as to the state of the audio engine, showed the Peak meter at maximum, while the Real Time meter hovered around the limit. Looking at the core usage in the htop interactive process viewer (https://htop.dev), it was interesting to note that only the performance cores were being throttled to around 85 to 91 percent; utilisation of the efficiency cores ranged from 14 to 28 percent.

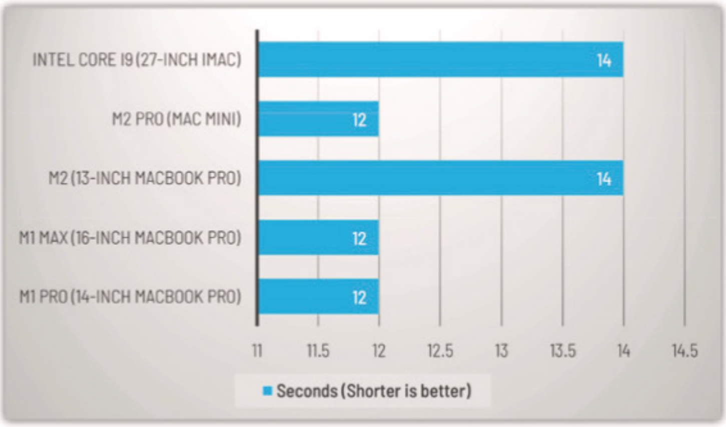
Enabling ASIO Guard, with the Level set to Normal, changed the system usage dramatically. The load on the performance cores reduced to between 64-78 percent usage, while the load on the efficiency cores increased to around 59 to 64 percent.

Steinberg don't explain the inner workings of ASIO Guard or detail how Cubase divides its workload between efficiency and performance cores, but these results suggest that ASIO Guard uses the efficiency cores to handle lookahead processing, whilst prioritising the performance cores for real-time work. This seems like an elegant way of exploiting Apple Silicon's heterogenous architecture in Cubase's audio engine.

Keeping the ASIO Guard Level set to Normal, the maximum track count increased to 60 tracks (and thus 120 instances of VST Amp Rack). Meanwhile, setting the Level to Low and then High resulted in track counts of 55 and 68 respectively.

One slight anomaly I discovered when I first started testing is that the Cubase.app Universal Application bundle still defaults to running the Intel binary under Rosetta translation when performing a new installation on a clean system. But this ultimately led to an unexpected observation, which was that my relatively basic test project performed better with Cubase running under Rosetta than natively under Apple Silicon.

However, rather than uncovering any weakness with Cubase's audio engine when running on Apple Silicon, what this anomaly illustrates is that VST Amp Rack isn't particularly well optimised for Apple Silicon. And this highlights an inconvenient truth: you can't just assume every plug-in from every developer will run more efficiently on newer, M-series Macs. Code optimisations made for Intel chips won't necessarily be equally efficient when running on a completely different architecture, and it would be relatively trivial to hand-optimise something for x86 that created a bottleneck on ARM and *vice versa*, particularly when implementing the type of convolution filters used by plug-ins like VST Amp Rack.



With Cubase 12's Mendel Bij De Leij mixed demo project, the differences in offline performance are less dramatic than for real-time playback.

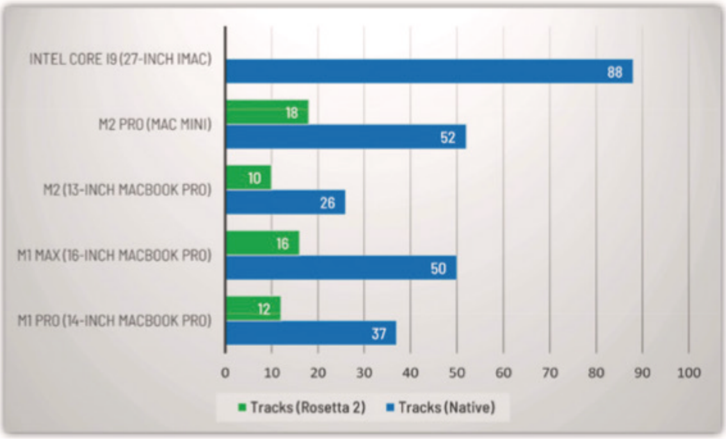
Concluding with an offline test, using Cubase's Audio Export function, I ran into some similar unexpected behaviour. Running either the Apple Silicon version of Cubase natively or the Intel binary under Rosetta 2, one of the two demo songs included with Cubase 12 — the Mixed version of the Mendek Bij De Leji demo project — took 12 seconds to export offline using either the M2 Pro-equipped Mac mini and 14-inch MacBook Pro, or my older M1 Pro-based MacBook Pro.

Pro Tools

In many ways, the success of Rosetta 2 — Apple's binary translation software allowing Intel applications and plug-ins to run out of the box on M-series Macs — has been both a blessing and a curse. It has allowed even demanding music and applications to run unbelievably well, with perhaps a 30 percent overhead compared with the same software running natively; but, in doing so, it has arguably permitted certain third-party developers to not make native Apple Silicon compatibility the highest development priority.

It took a while, but at the end of last year, Avid released a public beta of Pro Tools for Apple Silicon, with many of the usual caveats existing users will have previously faced in the adoption of newer systems. For example, many features aren't currently available, including MP4 Export, AAC Import, and the Score editor, while others will be no longer be supported, such as ICON, C24 and Control 8 control surfaces.

In terms of third-party compatibility, though, Avid have at least been



This chart shows the stereo track counts achievable by Apple's M1- and M2-based Macs (running Pro Tools natively and via Rosetta 2) and the last Intel-based Mac, with each track processing a single instance of Eleven Lite. It's interesting to note the Intel-based Mac handily beats the Apple Silicon models, due to the fact Eleven Lite runs more efficiently under Intel than ARM.

shepherding developers towards the goal of ensuring native AAX plug-ins are available. This means the vast majority of third-party AAX plug-ins can be used with Apple Silicon, as detailed on Avid's online Knowledge Base (https://avid.secure.force.com/pkb/articles/Compatibility/AAX-Silicon-Native-Plugin-Compatibility).

For testing, I once again recreated a similar Session to the examples used for Cubase and Logic Pro, although this time just using one instance of Eleven Lite (with the default preset) on a stereo audio track playing back the same file. With Pro Tools running natively on the M2 Pro-based Mac mini it was possible to play back 52 tracks simultaneously, a healthy improvement over the 18 tracks that could be played back in the same session with Pro Tools running under Rosetta 2. Still, 52 instances of a 'lite' version of a 14-year-old plug-in isn't a staggering result. Again, this points to Eleven Lite not being astonishingly efficient under ARM. And I don't think it's a coincidence that both Steinberg's VST Amp Rack and Eleven Lite make heavy use of convolution filters (and, unlike Logic Pro's Amp Designer, are required to be cross-platform).

Moving along to a real-world example, rather than use a demo song Avid include with Pro Tools, I decided instead to focus on one of my preferred 'real-world' test sessions: the final mix session of Netflix's *Sol Levante*, the first 4K HDR Atmos anime short. Netflix make the Pro Tools Session (including assets such as the video content) available as part of the company's Open-Source Content program (https://opencontent.netflix.com), which aims to provide a common reference for testing and prototyping new technologies.

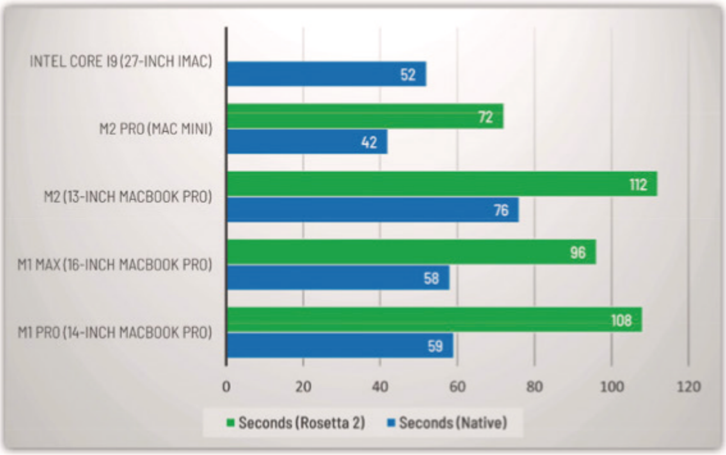
When I reviewed the Mac Studio in last year's May issue, I was dismayed that this otherwise extraordinarily powerful Mac was unable to play back the *Sol Levante* session without reducing the number of insert effects, although this was admittedly before the Apple Silicon version of Pro Tools was made available. So, I was simultaneously impressed and relieved to witness the new, M2 Pro-equipped Macs handle this session natively with ease, evn running alongside Dolby's Atmos Renderer (which still requires Rosetta 2).

As a further footnote to this performance, now that Pro Tools is running natively on Apple Silicon, a Mac Studio equipped with an M1 Max can now play back the *Sol Levante* final mix session just fine; and even the base, 13-inch MacBook Pro with an M2 and 8GB of unified memory can perform this feat. What a difference a year — or should that be, a native version of Pro Tools — makes!

An offline bounce of the 4:51 *Sol Levante* final mix session on the Mac mini with M2 Pro took just 42 seconds. This is a 40 percent improvement over the same bounce with Pro Tools running under Rosetta 2, which took 72 seconds.

Search For The Hero

Although the Intel-based Mac mini has remained useful, thanks to its upgradable memory and ability to boot Windows natively, by delivering the new Mac mini in two distinct M2-series configurations Apple have made it both more affordable and more powerful for those who don't require the pricier Mac Studio. The basic M2-based Mac mini starts at just £649 (£50 cheaper than before), bringing this particular Mac closer to its roots as "the most affordable Mac ever".



Shorter is better: the M2 Pro's additional CPU cores give it a distinct advantage when bouncing the *Sol Levante* final mix session offline. Given the real-world nature of this session, you can begin to see the advantages of Pro Tools running natively (as opposed to using Rosetta 2) on newer Apple Silicon-based Macs.

When it comes to choosing a MacBook Pro, after living with both 14- and 16-inch MacBook Pros for over a year, there's something about the 14-inch's form factor that I really like. The 16-inch has a desirably larger display, but the extra dimensions and mass make it less encouraging to carry around on a day-to-day basis. So, my own recommendation would be to go with the 16-inch MacBook Pro if you want a desktop replacement, but to choose the 14-inch model instead if you want something more comfortably portable.

However, while these new Mac mini and MacBook Pro models are highly desirable in almost every way, if I did have to muster some criticism, it would centre on two points. Firstly, as highlighted in last month's article, the performance of the internal storage used in the base models of both the Mac mini and MacBook Pro is a little disappointing. And the reason I mention this again is because the slower storage can adversely affect the overall system performance, particularly in models with less unified memory, meaning those with lower-end systems might have more of a business class experience rather than flying in first.

If one considers the price of the base Mac mini model, this is perhaps less of an issue, as you'd perhaps expect a few trade-offs to achieve the lower cost. But that this should be an issue with the base MacBook Pro model is a little harder to justify. And, on the subject of pricing, given that Apple Silicon Macs must be fully configured at the time of purchase, the cost of a system, including all the memory and internal storage you might require over the lifetime of the product, can quickly add up to an expensive purchase.

On balance, though, it's perhaps rather easy to take Apple's latest Mac mini and 14- and 16-inch MacBook Pros for granted. After all, the first round of Macs based on Apple Silicon made a profound impact, providing astonishing performance with an energy efficiency that took many by surprise. For creative professionals — and musicians and audio engineers in particular — everything about the M1-series Macs was just, well, better. And so, it's perhaps not surprising that the new M2-series Macs sensibly build on these foundations, refining, polishing, and pushing the performance envelope without sacrificing an economic use of power.

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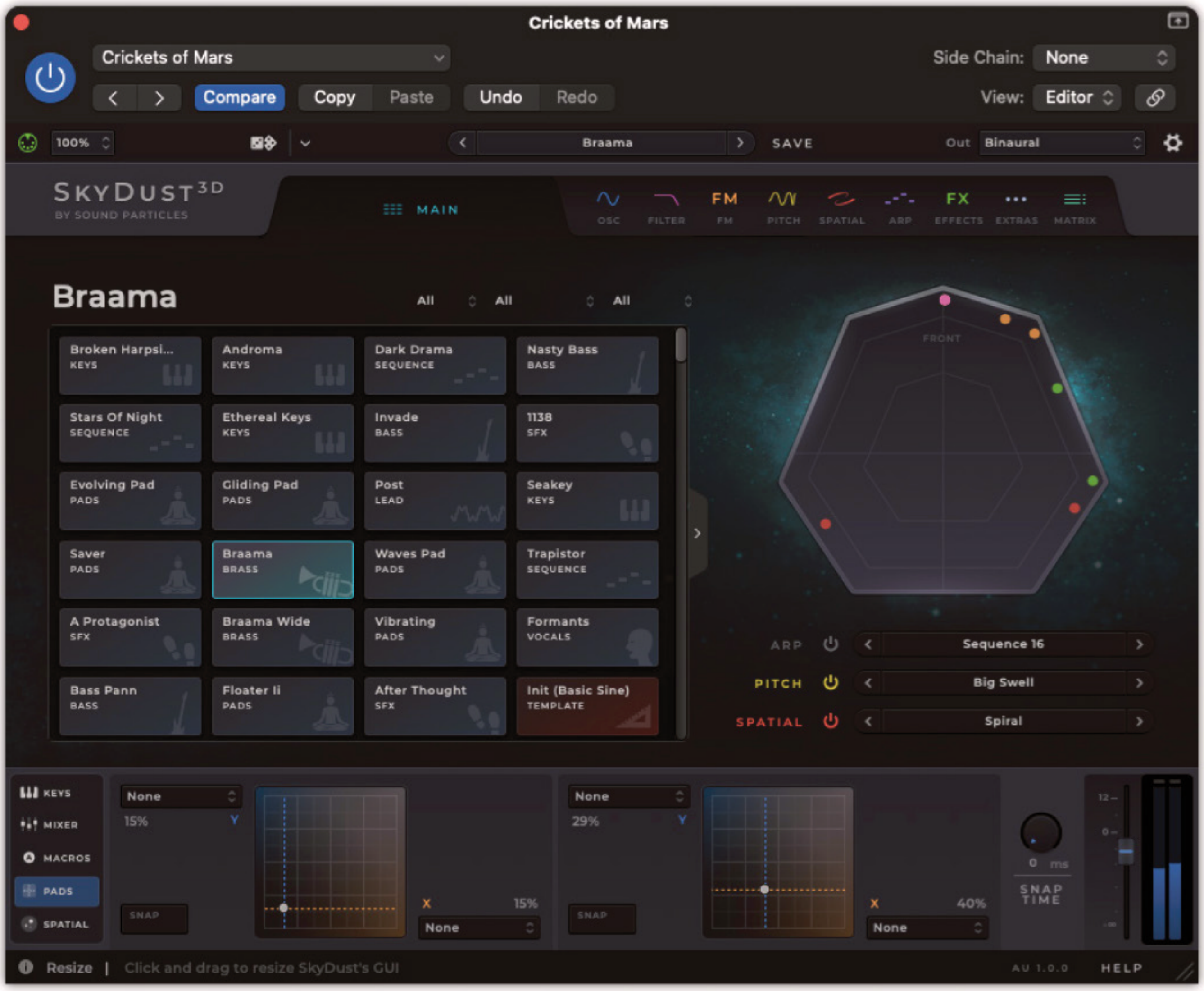
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Androma KEYS
Dark Drama SEQUENCE
Nasty Bass BASS
Stars Of Night SEQUENCE
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Evolving Pad PADS
Gliding Pad PADS
Post LEAD
Seakey KEYS
Saver PADS
Braama BRASS
Waves Pad PADS
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Formants VOCALS
Bass Pann BASS
Floater Ii PADS
After Thought SFX
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FRONT
KEYS
MIXER
MACROS
PADS
SPATIAL
SNAP
SNAP
SNAP TIME
0 ms
Resize
Click and drag to resize SkyDust's GUI
AU 1.0.0
HELP

Side Chain: None
View: Editor
Preset: Braama
Out: Binaural
ARP: Sequence 16
PITCH: Big Swell
SPATIAL: Spiral
None 15% Y
X 15% None
None 29% Y
X 40% None

Sound Particles SkyDust 3D

Software Synthesizer

Sound Particles' have placed 3D audio at the heart of their new soft synth.

PAUL WHITE

Sound Particles produce some very effective immersive sound processing software, and in their latest venture they've applied that technology to a synthesizer — which is apparently a world first. The synth itself uses a set of eight multi-waveform oscillators that can either be combined in layers for analogue-style subtractive synthesis or used individually as frequency modulation (FM) sources to modulate other oscillators. The oscillators are then fed into a mixer and each layer has its own filter, modulation and envelope settings. There are various options for further modifying the oscillator waveforms, and also onboard effects, arpeggiators and sequencers. Sound Particles have arranged the GUI using tabbed sections to make accessing all these features very straightforward.

The unusual part is that the surround panning technology that underpins most of the existing Sound Particles plug-ins has been integrated into the synth, so that surround pan position becomes a modulation destination. In other words, panning can be controlled by all the usual modulation sources such as envelopes, LFOs and sequencers, as well as having access to a set of preset trajectories. Each of the oscillator sources feeding into the mixer can have its own panning regime, so what you end up with is a multi-layered sound where the various layers each move independently. The output format can be selected to suit your monitor system from mono or stereo to a long list of almost 30 discrete surround formats up to 13.1 (and compatible with Dolby Atmos), binaural or Ambisonics, so there's scope for creating sounds that work on headphones as well as on surround speaker systems. Apparently the synth uses ACN/SN3D (otherwise known as AmbiX), so to use Ambisonics SkyDust 3D just needs to be placed on a DAW track that supports it. There are several binaural format options from which to choose, and further sub-choices for each of those formats. If fully immersive synth sounds don't fit into your workflow, then you can buy the less costly SkyDust Stereo, which has all the same features other than the output format options, which in that version are limited to stereo or binaural.

A large selection of patch presets is available and these can be viewed by category. There's also a Dice button to make random settings, with a menu that lets you lock specific aspects such as pitch, timbre or spatial settings so they won't change as you try different random settings. Presets are also available for arpeggiator sequences and 3D movement, so you don't have to do deep-dive editing until you are ready.

Views

The various views are selected via tabs at the top of the page, set out as Main, Oscillator, Filter, FM, Pitch, Spatial, Arpeggiator, Effects, Extras and Matrix. The Main page shows the factory presets as well as a graphical overview of the surround space, in which the different oscillator layers are represented by coloured dots. In addition to the main patch presets there are also presets for Arp, Pitch and Spatial settings, as well as a setting in the header bar for the output format. The oscillators offer a number of waveforms, adding noise and rectified waves to the usual analogue shapes. These waves may then be further modified by various folding, crushing and time-manipulation treatments. Pitch can be controlled by octaves, transposition and detuning. Each oscillator has its own envelope generator that can be selected from a simple AR type right up to a full DAHDSR (ADSR with added delay and hold). LFO modulation can

»

Sound Particles SkyDust 3D

£332

PROS

- Despite its apparent complexity the synth is very clearly laid out.
- A wide range of analogue and FM tones are possible.
- Unique ability to move each layer of sound within a virtual 3D space.
- If you don't need full surround support there is the option of a less costly stereo version.

CONS

- No facility to import user samples to use in place of the oscillators.

SUMMARY

SkyDust is a powerful synth in its own right even before the 3D aspects are brought into play. My own preference would have been also to allow user samples to be imported to replace the oscillators in one or more voices, but as things stand, the worlds of analogue and FM synthesis are combined in a very elegant manner providing huge scope for creative sound design. Bring in the 3D movements that can be applied to the individual voices and you have an instrument that is genuinely unique.

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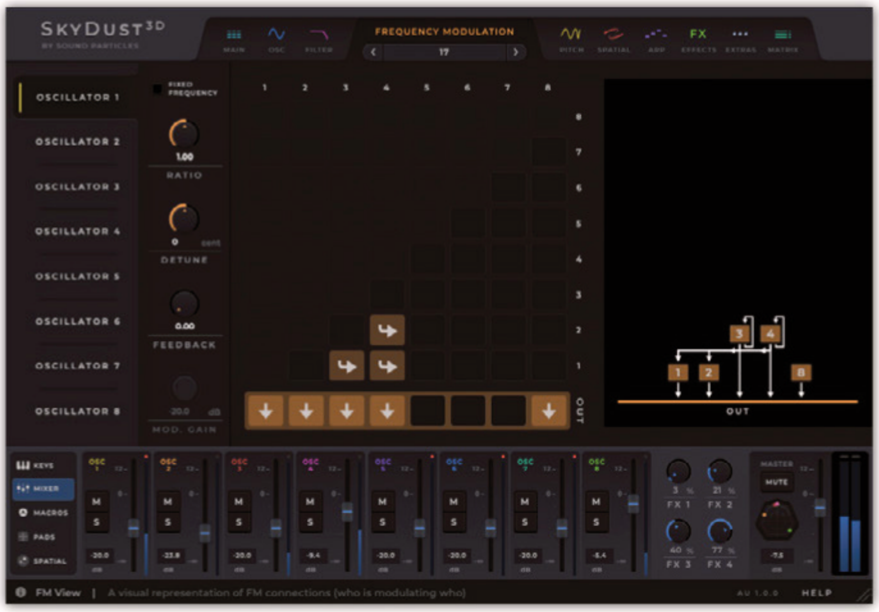
» be tempo sync'ed and also has a choice of waveforms.

Moving to the filter page brings up a comprehensive analogue filter type of arrangement with all the usual filter types and a choice of slopes, keyboard mapping, modulation velocity mapping and detailed envelope control. The FM page includes a matrix display as well as a block diagram of the oscillator connections, and it is possible to use any combination of 'straight' and frequency-modulated oscillators. Modulation oscillators can be routed to more than one carrier oscillator if required. The FM ratios can be adjusted along with detune, feedback and modulation amount. The Pitch page also offers envelope and multi-waveform LFO modulation with the addition of randomisation amount for specific parameters.

So far we're in reasonably familiar territory, but things change in the Spatial section, where movement in the 3D space can be controlled by LFO, sequencer, envelope or movement modifiers. These last are based on a selection of preset movement patches with definable start and end positions. Adjustment is also available for Azimuth, Elevation and Depth. Separate movement paths and rates can be set for note release phases.


SKYDUST 3D
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MAIN
OSC
FILTER
FREQUENCY MODULATION
17
PITCH
SPATIAL
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EXTRAS
MATRIX
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OSCILLATOR 2
OSCILLATOR 3
OSCILLATOR 4
OSCILLATOR 5
OSCILLATOR 6
OSCILLATOR 7
OSCILLATOR 8
FIXED FREQUENCY
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RATIO
0 cent
DETUNE
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FEEDBACK
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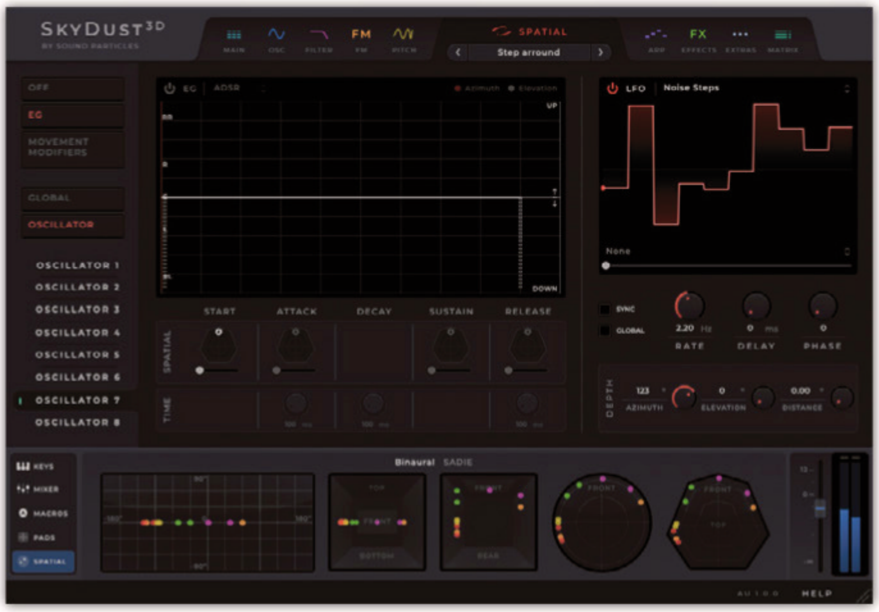
MOD. GAIN
OUT
KEYS
MIXER
MACROS
PADS
SPATIAL
FX 1
FX 2
FX 3
FX 4
MASTER
MUTE
HELP

FM View | A visual representation of FM connections (who is modulating who)

The FM screen is where the oscillators' routing can be viewed. Note that the Mixer view is open at the bottom of the screen.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OSCILLATOR 4
OSCILLATOR 5
OSCILLATOR 6
OSCILLATOR 7
OSCILLATOR 8
EG
ADSR
UP
DOWN
START
ATTACK
DECAY
SUSTAIN
RELEASE
SPATIAL
TIME
LFO
Noise Steps
None
SYNC
GLOBAL
2.20 Hz
RATE
0 ms
DELAY
0
PHASE
DEPTH
123
AZIMUTH
0
ELEVATION
0.00
DISTANCE
KEYS
MIXER
MACROS
PADS
SPATIAL
Binaural
SADIE
FRONT
TOP
BOTTOM
REAR
HELP

The Spatial view offers various views and control options for the 3D panning.

The arpeggiator has its own page and offers a choice of patterns, sync options, mono/poly note modes, scale quantise, number of steps and so on. Further choice is available in the Effects section where you'll find bit-crushers, delays, distortion, EQ and reverb. An Extras page provides eight further envelope generators and eight more LFOs that can be assigned to any parameter, while the Matrix page allows for setting up MIDI control and arranging the source and destination for the various modifier options. A Settings section accessed via the familiar cog icon accesses MIDI settings, polyphony, pitch-bend range, master tuning and so on, but you can also import an HRTF (Head Related Transfer Function) file in SOFA (Spatially Oriented Format for Acoustics) file format, in addition to choosing from the binaural presets available.

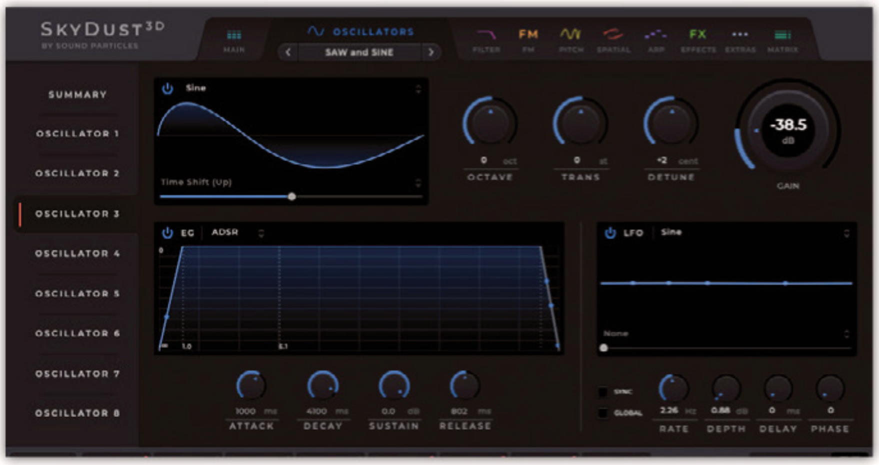
Bottom Panel

At the bottom left of the page are more tabs for Keys, Mixer, Macros, Pads and Spatial. In the Keys page you can activate Arp, Legato, Pitch and Spatial functions as well as seeing a smaller version of the surround space at the right-hand side. Mixer blends the various oscillator outputs and offers solo and mute buttons, rotary controls for four effects, and a meter fader accompanied by an even smaller view of the 3D space.

Macros allows multiple parameters to be assigned to the four Macro knobs, while Pads brings up two X/Y pads where the two axes can be assigned to various parameters. A snap mode allows the X/Y position to return to its original location when the mouse or trackpad is released. Spatial provides a number of different views of the voice movements along with a master multi-channel meter to show the levels of the voices being fed to the various surround channels.

Performance

Leaving aside the spatial aspects for a moment, the synth itself offers a combination of analogue and FM voices, which makes for a great deal of flexibility. Not being sample-based, you don't get the option of calling up specific instruments, but then that isn't the point of this type of synthesis. It is all about being creative, and there's plenty of scope for layering classic analogue sounds with tinkly or raspy FM sounds. There's a large range of presets to get you started, searchable by sound type, and here you'll find pads, basses,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MAIN
OSCILLATORS
SAW and SINE
FILTER
FM
PITCH
SPATIAL
ARP
FX
EFFECTS
EXTRAS
MATRIX
SUMMARY
OSCILLATOR 1
OSCILLATOR 2
OSCILLATOR 3
OSCILLATOR 4
OSCILLATOR 5
OSCILLATOR 6
OSCILLATOR 7
OSCILLATOR 8
Sine
Time Shift (Up)
0 oct
OCTAVE
0 st
TRANS
+2 cent
DETUNE
-38.5 dB
GAIN
EG
ADSR
LFO
Sine
None
1000 ms
ATTACK
4100 ms
DECAY
0.0 dB
SUSTAIN
802 ms
RELEASE
SYNC
GLOBAL
2.26 Hz
RATE
0.88 dB
DEPTH
0 ms
DELAY
0
PHASE

The Oscillator screen provides a choice of waveforms, modulation and envelope shaping.

synth strings, synth brass and things that go boing in the night. Adjusting the mixer offers a quick and simple way to tweak the presets, but it is also pretty straightforward to edit the envelopes and filters for each voice.

Setting up your own FM routing is a bit more challenging as there's a bit of a learning curve involved in getting FM to sound like anything other than collapsing scaffolding or a psycho dentist, but there's plenty of useful stuff in the presets section that you can use.

Space

The spatial aspect of this synth might seem like a gimmick but for certain types of music, including film work, it can be quite captivating as the voice layers that make up the various sounds move around independently of each other. We're already very familiar with stereo panning and other types of modulation that create a sense of movement, and SkyDust just brings all of that into the 3D world. As you might expect, such complexity takes up more CPU overhead the a typical analogue synth emulation, but given what's going on, I still found it to be very efficient.

While having the various voices buzzing around your head like flies might sound as though it would be distracting, in practice it generally feels more subtle, especially when playing polyphonically as the various movements blend to create a kind of 3D ensemble effect, with the various harmonics moving in space. Sometimes, when using faster movement settings, it can feel as though you have three or four Leslie speakers inside you head, but overall I found the effects to be musically enjoyable, and as the 3D aspects are fully editable, you don't have to go overboard with the surround effect if you don't want to. The spatial effects sound most effective on loudspeaker systems, which for most end users means a visit to the cinema, but the binaural options can also sound impressive over headphones while still sounding perfectly acceptable over loudspeakers, so that's the way I'd probably use the instrument in conventional music production. ■■■

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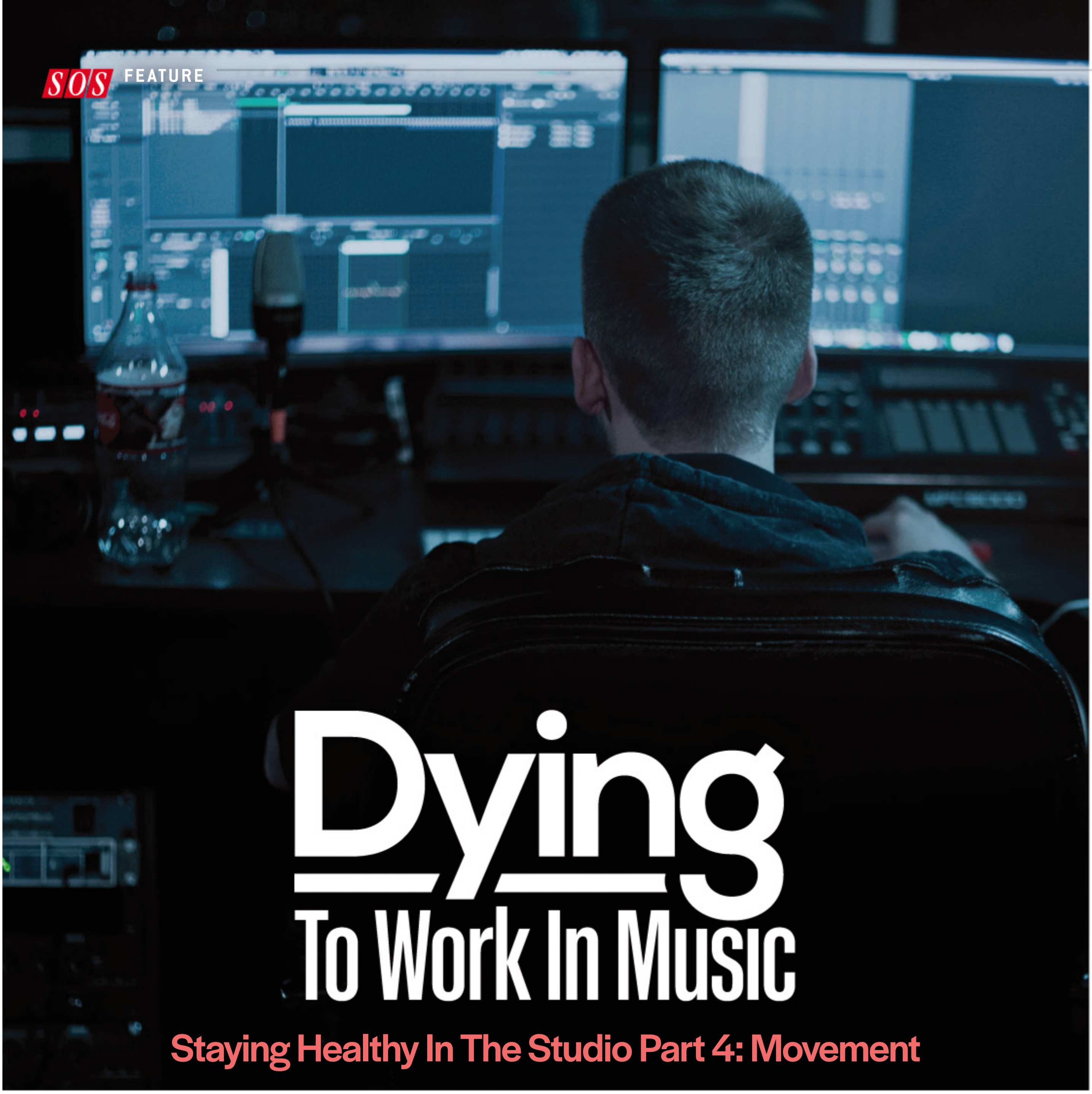

Occupations don't get much more sedentary than sitting at a console all day.

Dying To Work In Music

Staying Healthy In The Studio Part 4: Movement

When our job involves a lot of sitting down, it's up to us to inject some movement into our lives.

JACK RUSTON

"Twelve hours of sitting in a chair doesn't make a person more creative," says Tim Bazell. "We need to move, we need to be in the world, with our senses engaged."

"The long hours, inconsistent income and sedentary lifestyle can take a huge toll on your mental and physical health," agrees Sean Genockey.

In this series of articles, we're looking at the worrying decline in our baseline level of health, focusing on the obstacles presented by a job in the world of audio. The first three parts focused on sleep, stress and nutrition, and we'll conclude this month with movement.

While there are some setup days where studio engineers can be extremely active, never sitting down for a moment, for the most part that's not the case. As tracking gives way to overdubs, editing and mixing, the job tends to become increasingly sedentary. The engineer can bear the brunt of this, sometimes finding themselves at the desk for hours without a break.

Music & Movement

We hear a lot about the dangers of a sedentary lifestyle, but why does movement actually matter? It's all to do with the health of our skeletal muscle: the big motor units that we have under our conscious control. Not only do they move us around, but they play a crucial role in our metabolic health.

When our blood sugar rises after a meal, our muscle mass acts as a soak, topping up its stores of glycogen — the storage form of glucose. The more muscle we have, and the more we use that muscle, the better it can protect us against the harms of high blood sugar. When we move our muscle, we deplete glycogen and release myokines, signalling molecules that help with the uptake of glucose from the blood.

Conversely, if we're sedentary, our muscle mass shrinks, and begins to store fatty acids, which make it insulin-resistant. The inactive muscle refuses to take up blood glucose, which must be sent back to the liver where it can be turned into fat, packaged into lipoprotein particles, and sent to our adipose tissue. Over time, fat begins to accumulate in the liver itself in a condition called Non-Alcoholic Fatty Liver Disease (NAFLD). The liver in turn becomes insulin-resistant: it stops listening to insulin's message that there's plenty of glucose already available in the blood. It continues to release its own stores of glucose, breaking down muscle tissue to create more.

It's a perfect storm. What we need is more movement, more muscle mass and less surplus blood glucose, but what we wind up with is muscle being broken down to fund the creation of blood sugar that we don't need. This is why diabetics have high blood sugar, even when they haven't yet eaten. Non-Alcoholic Fatty Liver Disease, which is worsened by the excessive consumption of fructose from soft drinks and processed food, represents a crucial stepping stone in the path to Type 2 diabetes. It affects nearly 40 percent of adults worldwide, and shockingly, 5-10 percent of children in the US.

Stress makes this whole cascade worse as cortisol affects our insulin sensitivity. We start to see symptoms like irritability, weakness and shakiness when a meal is delayed or missed, tiredness, cravings for sugary or salty foods, fatigue in the afternoons and fat around our mid section and face. These indicators, which we so readily accept as normal, are a red flag: a warning that our metabolic health is on the ropes.

The 'E' Word

Luckily, this is something that isn't terribly difficult to improve. We talked last month about getting rid of processed food and drinks, and now we turn our attention to movement. When we move our skeletal muscle, we prevent that storage of fat in the muscle mass, deplete glycogen reserves and so increase our capacity to store blood sugar. If we can move in such a way that actually builds more muscle, we further increase that capacity, as well as increasing our basal metabolic rate: the number of calories we burn just sitting in front of the console, or at the computer.

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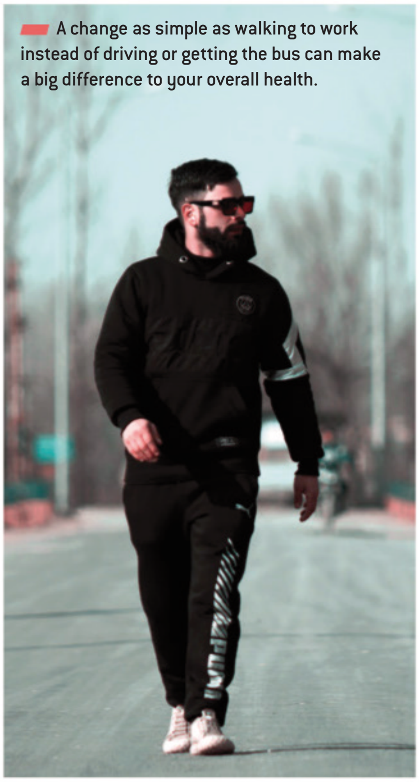
A change as simple as walking to work instead of driving or getting the bus can make a big difference to your overall health.

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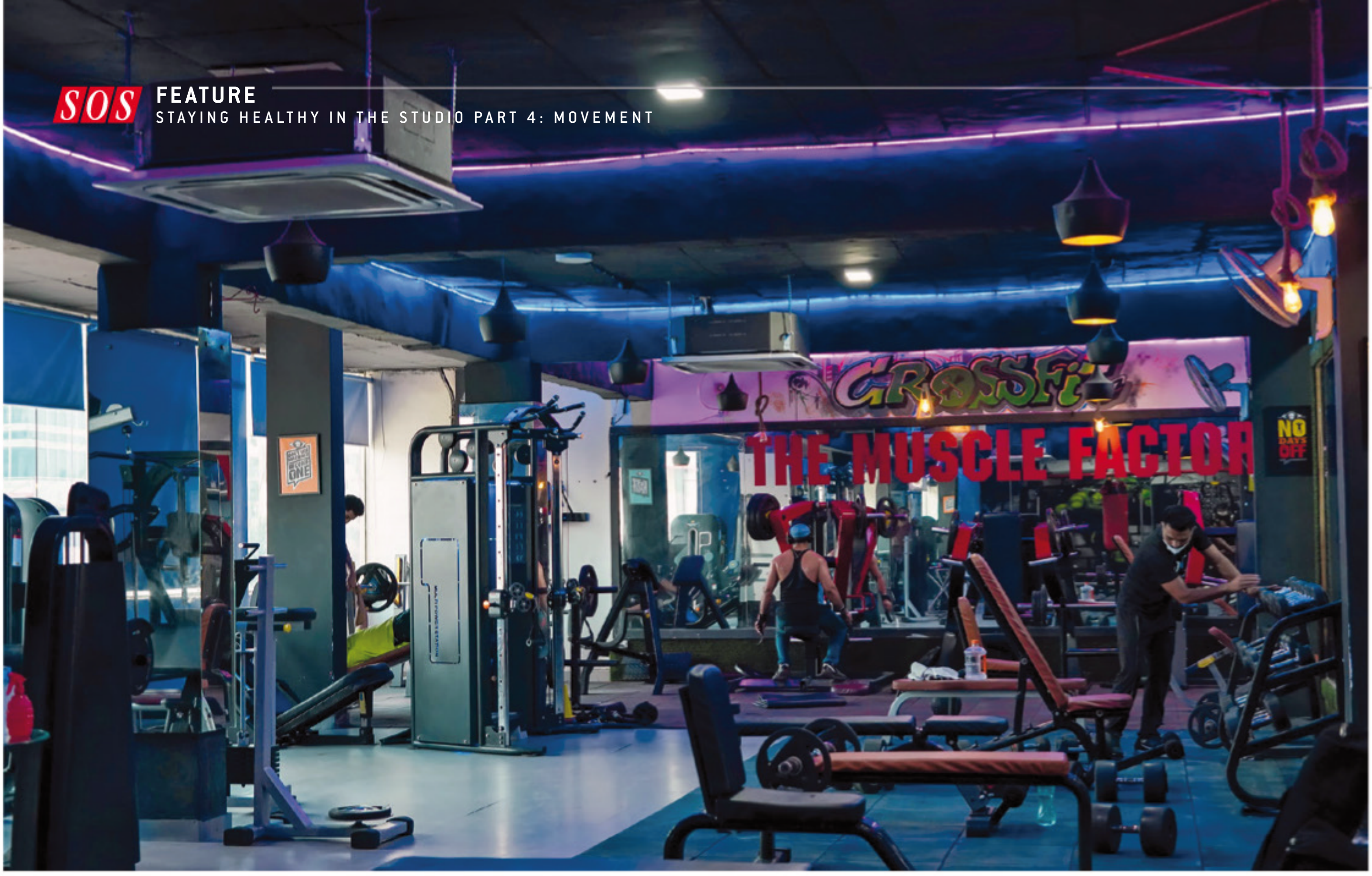

» In all this discussion of movement, you may notice that I'm avoiding the word 'exercise', the connotations of which are horrible. To many of us, 'exercise' means hours of arduous torture on machines that lie to you about how many calories you've burned, attempts to watch Netflix on a tiny phone screen with your noise-cancelling AirPods no match for the PA blasting Skrillex and Eminem, and the Myzone heart rate display on the gym wall detailing the infinitely superior efforts of your fellow cardio bunnies. Perhaps you eschew the gym entirely for brutal early-morning runs on knees that seem to be designed for almost anything else. If you're getting a sinking feeling just thinking about it, quite right — it's all awful.

Some of us love running, and I'm not necessarily trying to discourage that. Indeed, if we really want to be fit in every possible sense, to maximise every benefit that exercise has to offer the human body, we will need to include cardiovascular exercise at a variety of intensities, from endurance work all the way through to very high-intensity, 'best effort' interval training. That doesn't mean we have to run, but it does mean that we'd be spending at least some of our time undergoing what most of us think of as 'exercise'. If we're lucky we can mask the misery by doing this in the form of a sport or game that we enjoy. But here's my point: If our time is limited, if we're not in love with idea of sweating our lives away on the treadmill, and if we're simply not going to do this if that's what it involves, there are ways we can move which will give us more 'bang for the buck', more return on our investment of time and effort.

Make Certain

The recommendations made in this article will benefit the large majority of people, but be aware that there may be reasons why some of us should not exercise in certain ways, or at certain intensities. We recommend that you get the go-ahead from your doctor before undertaking an exercise program.

The first thing is to understand our goals. We are looking to break up the sedentary hours sat at the computer, to move frequently throughout the day, and to maintain the volume and function of our lean body mass — those parts of us that are not devoted to fat storage. The problem with endurance running is that it tends to confine our daily movement to one condensed period of time, when what we really need is consistent, regular movement throughout the day. Endurance exercise also drives an adaptation that favours efficiency and a reduction in muscle mass. Long-distance runners are skinny. What we need, what will suit our requirements is to actually be less efficient with energy, and to build more metabolically active lean mass.

So if not running, then what?

Walk!

Walk for transport. Walk immediately after meals. Walk for 10 minutes. Walk around the block. Walk and talk. Walk if you need to make a call. Walk to clear your head. Walk to rest your ears. Just taking three 10-minute walks per day, after meals, clocks up nearly 550 miles per year! If you're sedentary at the moment, and you do this and only this, you're making a huge difference to your health.

Walking proved to be an essential activity for many of the engineers and producers I spoke to. Jim Homes states: "If a band is feeling lethargic, we take them for a walk around the block. It's a much better dopamine hit than a bar of Cadbury's. We take time away from the studio as much as we can afford to. We can get out into nature very easily and we do, all the time. It's powerful."

Sean Genockey walks every day, and Emre Ramazanoglu explains: "I take constant breaks, and try to go outside at least a couple of times a day. I walk to work, four to six miles each way depending on my route."

We don't need to cover 12 miles a day if we don't want to, though. The typical 10,000-step goal is a good one. In fact a recent study identified optimal outcomes at 8800 steps per day.

Lift Heavy Things

If we're after a form of movement that takes little time, builds and maintains muscle, protects our metabolic health and increases our mobility, function and physical robustness as we move past middle age, we need look no further than strength training. Even 30 minutes of resistance exercise, twice a week, is enough to improve every marker of metabolic health. To protect us from age-related muscle loss and osteoporosis, we need to build lean mass, and we need to do it now.

Gyms, and their users, come in all shapes and sizes. Don't be intimidated!

I fully appreciate that there are barriers to beginning a strength training practice for the first time. Some of us are not going to take that step, regardless of how much it might benefit us. While I'd strongly encourage anyone to give it a try before dismissing the idea entirely, if walking is what you're able to do, that's a huge step (no pun intended). Other than simply not wanting to, what are our barriers to entry when it comes to strength training, and what can we do about them?

Better, Not Bigger

Many people, especially women, are worried that strength training may make them 'bulky'. While it's encouraging to see that the painfully thin 'size zero' imagery that has caused such harm over recent decades is slowly starting to be balanced by stronger-looking phenotypes, it's still the case that many of us would prefer to be physically small. Rest assured that by doing strength training, by increasing lean mass, we are likely to get stronger and smaller: we lose inches as our body fat percentage drops. It may be that our scale weight increases as bone density and muscle mass increase, but we will typically shrink. The idea that we will pile on a significant volume of visible muscle is a myth: It's incredibly hard to build a lot of muscle, and we're not going to do it by accident. Even men, with their relatively greater propensity for anabolism, will only gain a lot if they're using drugs or are particularly genetically gifted. In the case of really huge people, it's both. Don't confuse the 'pump' that you might experience for a short time after training for a permanent increase in muscle size.

We're going to need to join a gym, and to learn some new skills. Some of us will need to overcome our preconceptions about gyms, about the judgement we believe we will be subjected to there. Many of us don't see ourselves as 'gym people'. I've been training in commercial gyms for decades, everything from expensive London 'health clubs' to boxing gyms where most of the weights look like they've been put together from bits of old cars. In that time, I've found it's extremely rare for anyone to look down on a fellow member for their lack of fitness, ability or physical condition. It tends to be the largest, scariest individuals who most fiercely defend the right of anyone and everyone to use the gym, and who most generously give their time if ever asked for help. The less comfortable someone appears to be at first, the more they will be respected for making the effort.

Men tend to be less strong than they think they are. Women tend to be far, far stronger than they think they are. There's no point in measuring ourselves against others in this context. With the exception of two people in the world, we are all stronger than someone, and weaker than someone. In my case, the person I'm weaker than often proves to be half my size, and a girl. The only comparison worth making is you today vs you yesterday.

Strength training need not be time-consuming. We are not trying to burn calories, but rather to send a message to our central nervous system. It's a message that says: "I need to be strong, I need more muscle, and I need it to work well." The best way to send this message in a short time is to lift fairly heavy, focusing on a small number of big, compound movements — those that use multiple joints. We may need some help to learn correct technique for these lifts, but these are important skills that will serve us for the rest of our lives. There are people in their 80s performing these lifts in my local gym. They look like they're 65.

The best gym is the one you can get to. It's no good joining an incredible gym that's so far away that you can't make it over there. Find your nearest gym and go in. Get the tour. Just see what it's all about. If you have a choice of gyms within easy reach, look for one that seems to prioritise strength work. Do they have a lot of squat racks, barbells, and lifting platforms, rather than acres of cardio machines? Is it full of the sort of people that look like they strength train at a high level? If so, great.

Strength work can sometimes be intense, but much of the time we are simply popping into the gym to practise our lifts for a little while. We are often playing, enjoying what our bodies are capable of, enjoying making them move. This need not be sweaty and arduous. It can be fun!

Stay Well

This series is not intended to be definitive, or exhaustive. I hope that it represents the opening of an expanding conversation, as more industry professionals and organisations engage to offer their perspectives. In the meantime, as Jerry Springer would say, look after yourselves, and each other. ■■■

NEW HARMONICS

The Art Of Audio Fermentation

We look at the processes that can add harmonic content to a sound, for subtle warmth or radical transformation.

OLI FREKE

You might see a sound engineer as an audio chef, blending the finest ingredients to create the perfect sonic dish. But there's another food analogy that I think is interesting to consider, that of fermentation. Fermentation is the use of organisms such as yeast and bacteria to transform otherwise innocuous foods into more complex and interesting ones such as bread, beer and chocolate. Similarly, certain audio processes can introduce new frequencies and harmonics that weren't there before, and which can create spicy new audio flavours and tasty sonic experiences.

Techniques such as distortion, saturation, clipping, resonant feedback and waveshaping are all methods of introducing brand-new harmonics into a sound. This article will look at a few of the ways to 'ferment' your audio in this way, introduce new harmonics and to creatively transform its character.

Distortion Overload

The most common method of adding harmonic content to a signal is through distortion. Technically speaking, distortion is a description of what happens when a signal is transformed in a non-linear way as it passes through a system or a component. Which is a technical-sounding term for a relatively straightforward idea, as shown in Diagram 1.

A non-linear response introduces harmonic content: feed a sine wave in, for example, and you may get additional sine waves produced a fifth or an octave above. This is normally undesirable, and equipment designers vie to achieve the lowest Total Harmonic Distortion, a measurement of how much unwanted distortion is being introduced.

However, it was realised quite early in the history of recording that certain kinds of distortion actually sounded good, and enhanced the power of the music. All kinds of distortion were experimented with, from 'quite subtle' to rock music's full distorted guitar sound, which opened pathways to whole new musical genres.

Brian Eno has written that by distorting a sound, one is subconsciously stating that the music is bigger than the system can even contain, implying its power and might! Which certainly works for rock music, but there are different ways of introducing distortion to a sound and they don't all have to be as obvious as the kind you get from a fuzz pedal or distorted guitar amp. Applying distortion more subtly can give us the benefits of new harmonics, warmth and a richer tone without totally transforming the signal.

Amplitudinal Clippage

The easiest way to get a non-linear and hence distorted response from a component or device is to feed it too high a signal level. In this state, it won't be able to represent the signal accurately, and eventually, we'll reach the point where it simply can't produce any more voltage, no matter how much we pile on the input signal. This state is known as clipping, because the top and/or bottom of the waveform gets flattened out.

Synth aficionados will realise that as a signal becomes progressively more clipped, it starts to resemble a square wave. And as a square wave is the theoretical sum of an infinite number of odd-numbered harmonics, it's no surprise to find that clipping introduces predominantly odd-numbered harmonics. In extremis, it can actually turn a guitar or bass note into something that sounds a lot like a synth square wave. It also means, interestingly, that in theory, clipping a square wave has no audible effect! In practice, hardware and software clipping do other things to the signal which may indeed alter the sound further.

When clipping starts abruptly at a set level it's referred to as 'hard clipping'. Signals just below the threshold are completely undistorted, while those just over it are clipped. It's not the most subtle way to introduce distortion, and it's not typical of most analogue circuits. More often, things become progressively more non-linear above a certain level, so you can get some distortion before full clipping is reached. This response is referred to as 'soft clipping'. ('Hard-knee' and 'soft-knee' settings are similar concepts found in compressors).

Each piece of equipment is different, of course, and will respond differently to different amplitudes and types of material in relation to their thresholds and non-linear response curves, and so will generate different sets of odd or even harmonics at different rates and intensities. Software emulation of distortion can take this kind of manipulation into very sophisticated realms: dynamic clipping thresholds, variable frequency responses, parallel distortion channels, and so on.

Worth noting is that some classic guitar fuzzboxes use a distortion curve that's asymmetric. This means the positive-going and negative-going halves of the signal are distorted differently, and usually causes the addition of even-order harmonics to the already present odd-order harmonics. This manifests itself as a generally fuller tone, with more low end. Symmetric curves, by contrast, generally only generate odd-order harmonics, which is sometimes better when an already full sound doesn't need the full odd and even treatment!

Altering the EQ of a signal going into a distortion unit can have quite a significant effect on the distortion behaviour, and is always worth exploring. The low end will usually be driven into distortion more quickly, as it's where most of the energy lies, so if that bass is distorting too much and overwhelming the rest of the sound, cutting low end before the distortion unit can help balance the resulting distorted tone.

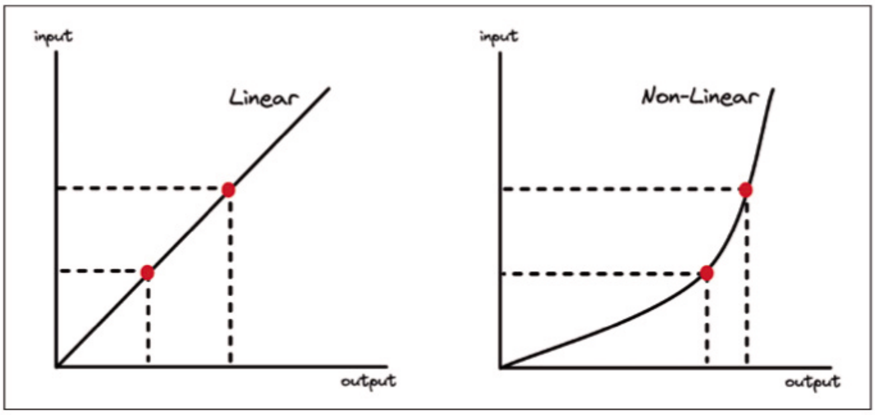

Diagram 1: 'Linear' just means the signal passes from the input to the output without change: the mapping from input to output is one-to-one, creating a straight line, hence linear. So 'non-linear' just means the line is no longer straight, representing the fact that the signal has lost that one-to-one mapping between input and output, ie. it's been changed.

Clipping distortion works well on single instruments as a creative tool. It can produce a thicker, richer tone, with increased high-frequency content. By the same token, though, it's often too much on more complex material as every frequency present creates its own set of harmonics, along with all the others. So much high-frequency content is then generated that it tends towards white noise and send the whole mix into meltdown. For subtler use cases, such as on groups of instruments, mix busses and whole mixes, the gentler type of distortion described as saturation is usually more useful.

Saturation

Saturation, in the analogue realm, occurs when the signal starts to exceed the level at which a device can represent it in linear fashion. Signal peaks are not clipped, but neither are they represented at their true amplitude. Sometimes the saturation effect is achieved by deliberately overloading a mixing desk, tape machine or tube compressor, but there are also lots of hardware and software products specifically designed to introduce pleasing saturation. As well as sympathetic harmonic content, saturation also reduces the dynamic range of the signal. Transients may also be softened, because these are often the biggest peaks in the signal and so cannot be represented accurately by the component that's being saturated. Used carefully on a mix bus, saturation can provide that 'glued' feeling thanks to the compressing effect, and add richness from the added harmonics.

By modifying the frequency range of the signal going into a saturator, we can create an enhancer: an alternative to high-frequency EQ that adds harmonic content to the upper frequencies to 'excite' the mix. Try boosting a small frequency range with EQ before the distortion effect, then apply the opposite EQ curve to the output signal. The chosen frequency range won't be louder than before, because the bulk of the new harmonics added by the distortion will relate to that region.

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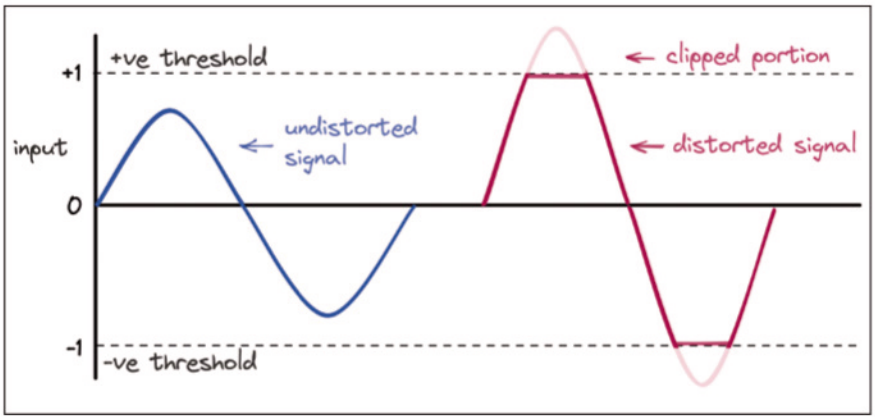

Diagram 2: Clipping is easily illustrated using the example of a sine wave. As the top and bottom of the waveform are 'clipped off' it becomes more and more distorted, tending towards a square wave.

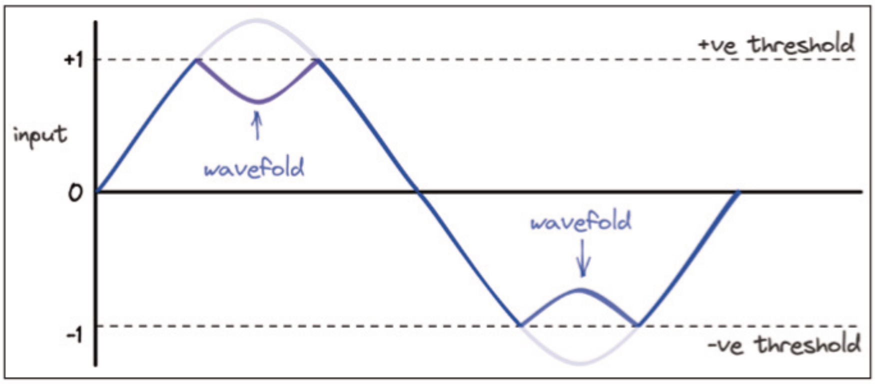

Diagram 3: The simplest form of wave-shaping to illustrate is the wave-fold; ‘reflecting’ back the top of the waveform above the threshold. This introduces all manner of related harmonics. It’s even more effective when the threshold, bias or phase are changed dynamically for a really exciting sound.

» One classic use case is to make a subby bass more audible on smaller speakers by adding distortion to the upper frequencies, filling out the low midrange so that the bass seems louder even though the mix level hasn’t been changed. This can be done through a selective EQ parallel distortion process: create a parallel channel of the bass with plenty of distortion (to taste), cut the low end of this distorted signal so the original low end of the bass is unaffected, but bring in the top end of the distortion over the old signal to boost the higher mids and tops.

In the real world (and in emulations of it) there are some common sources of saturation, such as magnetic tape, tubes, valves and transformers, each of which has its own characteristic response to being pushed beyond its linear range of operation.

Audio tape generates both odd and even harmonics as it runs out of magnetic particles to represent increases in signal level. Interestingly, the level of third-order harmonics (third-harmonic distortion) was used to calibrate tape machines, and usually at 3 percent (which is a pretty high level of distortion). An overdriven tube is often particularly rich in second-order harmonics, and because these second-order harmonics are relatively lower in frequency than the other upper-order harmonics, this can sound warmer than other types of saturation.

Saturation can be added when tracking instruments by overdriving the mixer channel or mic preamp, potentially pushing transformers and other components outside their linear range of operation. This is a much-used technique, but needs to be done carefully, as the saturation can’t be removed afterwards. Overdriving mixer channels at mixdown is another method of achieving a saturated sound, and saturation across an instrument group or bus will give that glued feeling, due to the shared compression effect and harmonic generation. When used across a whole mixdown in mastering, it will generally need to be a lot subtler and treated more as an enhancement rather than an attention-drawing ‘effect’.

Wave-folding

Although distortion and saturation of these kinds are probably the most common ways of adding new harmonics to a signal, they’re not the only ones. Synthesizers have had a method called ‘wave-folding’ since the late 1960s, and this was used on the earliest Buchla and Serge modular systems.

Instead of clipping the amplitude of a signal, they ‘reflect’ the waveform back above a threshold by inverting it. Add phase-shifts and bias control over the precise nature of the wave-fold, and they can create a huge range of wild and exciting sounds. With the explosion of Eurorack modules over the last 20 years, it’s become a much less niche method and is part of the new vocabulary of synthesis. Frequency modulation (FM) and amplitude modulation (AM) are further methods of cross-pollinating audio signals to create new harmonics.

Bit-crushing

Distortion doesn’t have to be added in the analogue domain, either. By reducing either or both of the word length and sample rate of a digital signal, one can add another style of crunchy old-school digital distortion. Unlike the odd- and even-order harmonics added by analogue distortion, the distortion products of bit-crushing tend to be unrelated harmonically to the source signal. From a sonic point of view this makes them much grittier or harsher, which is great for sound design, perhaps less so for ‘warming up’ a mix! It’s a commonly found effect in modern DAWs and is often effectively used on drum loops and percussion samples to ‘rough’ them up a bit.

In Summary

So, to return to the culinary analogy, we don’t have to limit ourselves to blending a few basic ingredients or feel stuck with their current nature, we can be audio fermenters, devising new ways to introduce extra flavours into our audio bakes and surprising ourselves — and our audience — with newfound harmonic content, and sometimes brand new genres of music into the bargain! Hopefully the examples discussed here will give you food for thought and some ideas to explore in your own sonic kitchen... ▮▮▮

Modern Tools

A lot of the discussion in this article has covered how analogue equipment generates saturation and distortion, but there are many (many) software options that can get close to the real-world effect. All modern DAWs come with an excellent range of distortion, overdrive, saturation and bit-crushing effects, which will provide 95 percent of what most of us would reasonably need.

And of course, in addition, there are a multitude of free and paid-for third-party plug-in options too. It’s hard to recommend any particular one over any other, so choosing according to taste and need is the order of the day. But hopefully now, with a deeper understanding of what these tools are modelling and what part they played historically, it’s easier to see which type might be best suited as a starting point in your own work.

Of course, there are still plenty of analogue options available, including — covered in April’s *Sound On Sound* — an intriguing new ‘harmonic EQ’ device by Cranborne Audio, called the Carnaby Harmonic EQ, which instead of simply boosting an EQ band, introduces tuned harmonic distortion in that range instead. Intriguing!

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AIR Music Technology AIR Delay Pro

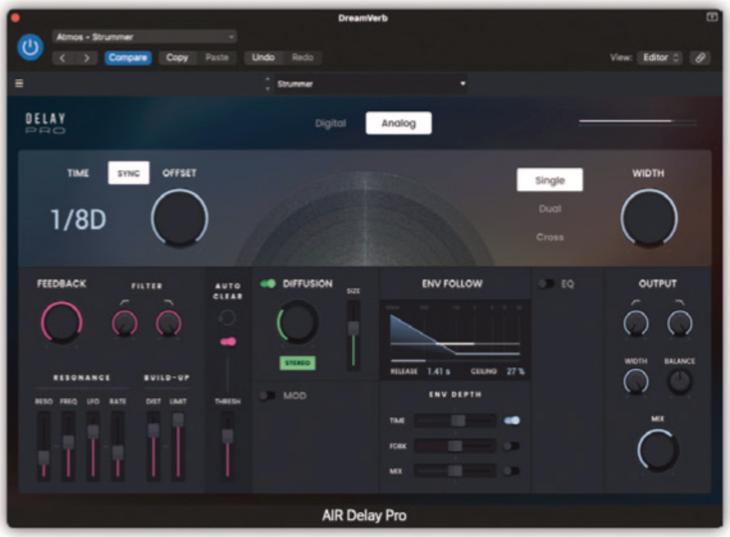
Delay Plug-in

The world may be awash with delay plug-ins but in AIR Delay Pro (available in all the usual Mac and Windows plug-in formats), the AIR Music Technology designers have delivered some interesting twists that make it worth a closer look. Offering both digital and emulated analogue delays in a dual-delay format, AIR Delay Pro also has onboard resonant filtering for the feedback path and modulation of the delay time, and the modulation type can be Wobble, Chorus or Stereo. Both high- and low-cut filters are available in the feedback section, as well as distortion and limiting. A Diffusion section, with variable stereo width, also allows for a little reverb-like 'blurring' to be added around the repeats.

EQ, just to the left of the output section, offers frequency, gain and Q (bandwidth) controls while the output section itself has further high- and low-cut controls, in addition to Balance, which adjusts the pan position of the delayed signal. A Width control affects the stereo delays, and another called Mix sets the wet/dry balance in the usual way.

But what really sets this delay apart from the crowd is its envelope follower. This has a variable release time, and its output can, using a threshold control, be used to modulate the delay time, feedback and wet/dry mix. Each of these destinations has a separate amount slider and an on/off switch too.

The delay modes on offer are called Single, Dual and Cross. In Dual mode the two delay times can be set independently either manually or in synchronised subdivisions of the host tempo, whereas in Cross mode they're linked. One unusual feature is Auto Clear, a threshold-based process that clears the delay feedback buffers when the feedback signal drops below the threshold level set by the user — a lot of the more unusual effects come courtesy of this feature.



Taken in combination, these features extend the creative capabilities of the delay effect significantly — well beyond what a typical delay can do. Of course, you *can* get all the usual flavours of delay, in which case filtering and modulation can conjure up some very tape-like timbres. But that clever Auto Clear feature, and the way the various parameters can be linked to the envelope follower, open up the possibility of creating effects that react to the dynamics of the music, including adding pitch swoops, dynamic filter changes, filtered rhythmic repeats, exotic dub effects and so on. Many of the less usual effects come courtesy of that Auto Clear function, which can produce some quite surreal results when the feedback level is set high.

To give you an idea of the breadth of effects possible, it's worth exploring the generous preset collection, which is arranged by category. The more you dig in, the more creative potential you discover. Maybe not every pop song composer will push the limits of AIR Delay Pro but if you're into EDM, chillout or dub, you're going to love it, and there is a free demo period if you'd like to take it for a test drive first. *Paul White*

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The Creative Electronic Music Producer

Book Review

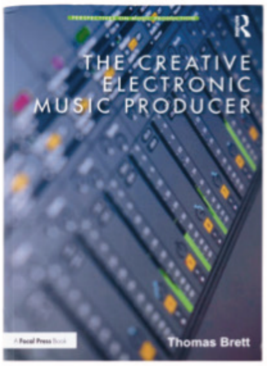
Most *Sound On Sound* readers probably have some level of interest in electronic music, so Thomas Brett's book (published 2021 by Routledge/Focal Press, ISBN: ISBN 978-0-367-90079-3) about creating and producing electronic music is likely to be of interest. The author is both a professional percussionist and an electronic music producer, having released several of his own recordings involving tuned percussion and electronic sounds, as well as publishing compositions specifically for marimba and vibraphone.

This is not a step-by-step 'how-to' book of driving a DAW or synthesizer, though. Naturally, there are practical aspects and advice aplenty too, but Brett's approach is a cerebral one — a deep philosophical analysis of the various creative inspirations, ideas and processes involved in producing electronic music, so for the reader this manuscript is definitely more of an intellectual exercise than a practical one.

The book's 156-pages are divided into eight main chapters, plus an introduction and an appendix, which lists a very eclectic selection of recommended listening material pertaining to each chapter's contents. An introduction explains the author's own inspirations, history and personal approach to electronic music creation and production, circling around the fundamental idea that producing electronic music is open-ended; it's a game of infinite possibilities of recording, sampling, synthesizing, sequencing and arranging, all with myriad options. Throughout the book Brett quotes a multitude of established electronic music composers, drawing wisdom from their comments, which are mostly derived from interviews and articles published across a wide range of sources (all meticulously listed in a References section of the book).

Chapter one examines the three supporting pillars of electronic music production (sampling, MIDI and the DAW), with a brief history of their developments and the resulting capability of working entirely ‘in the box’ should that be appropriate, culminating in a case study using Ableton Live. The following chapter moves on to production workflows and how software and hardware are integrated to form a ‘system’. Brett then considers how the producer steers the music direction while also being steered by the system itself as it inspires ideas through its own complexities. Brian Eno’s generative loop machines idea is used as a case study.

In the next chapter Brett describes how to start: the ideas of beginning anywhere and of improvising, of working quickly to capture ideas before overthinking them, and of using placeholder chords to provide a shape without having to delve into detail. The same broad concepts apply to the selection of sounds, which is the subject of chapter four, looking at the role of presets to access sounds quickly and to create palettes of complementary sounds. This leads on to the history of sampling systems, the whole idea of sound design, and of synthesized versus natural organic sounds. Chapter five considers the role of YouTube for sharing work with peers and fans alike, and for its countless production tutorials and demonstrations, with a group of interesting case studies of how these can be used in your own music production processes. Next up is a chapter on rhythm programming, the anatomy of a groove, and Brett’s five rhythmic principles.



Chapter six looks at the role of disruption in moving the production process forward, re-energising the interest, and to avoid things sounding predictable: how mistakes can be folded back in to the music as a modulation to add new interest and to spur new ideas. The same goes for re-sampling sounds within a track to use in layers elsewhere.

The penultimate chapter explores editing as a production tool, adjusting phrasing, and creating repetitions and variations. Lastly, Brett delivers a chapter on arranging and mixing, and describes a variety of approaches and techniques, again with fascinating case studies. In his conclusion Brett lists six key production concepts: contrast, compensations, musical lines, surprise, efficiency, accumulations. And if you want them explained you’ll need to read the book!

I am, fundamentally, an engineer with a very practical root to my own electronic music making, but Brett’s book has given me lots to think about in terms of my intellectual approach, my inspirations, and my processes. The many fascinating insights into the methodologies of the greats of electronic music production have made *The Creative Electronic Music Producer* an inspirational and motivating read. *Hugh Robjohns*

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Electro-Harmonix Lizard Queen

Octave Distortion Pedal

Designed in collaboration with the JHS pedals team, EHX's Lizard Queen is a slightly unusual distortion pedal — there's no control for regulating the actual amount of distortion other than the volume control on your guitar! Housed in a compact pedal format and powered by either battery or an optional external supply, the pedal takes only 5mA, so battery life should be long. Other than the bypass footswitch there are just three controls, labelled Volume, Octave and Balance. The octave effect created here isn't as obvious or as dissonant as you might get from something like an Octavia fuzz pedal, but if you are planning to include a cover of Purple Haze in your set it will get you close enough.

Volume does as you'd expect, and is used to produce an appropriate overall effect level relative to the bypassed level. Octave brings in more of the octave effect as it is turned up, which in this type of pedal is generally created by full-wave rectification of the original signal, to create a distorted upper octave. Balance adds a light warmth to the output when turned anticlockwise, or a little more edge when fully clockwise, the idea being that you can emulate the sound of a warm and cuddly fuzz/distortion box or dial in something more raspy. This control has a fairly subtle effect and is most obvious when the octave sound is added.

The basic distortion sound is well chosen, given that its amount is preset: it delivers a convincing Billy Gibbons type of grind that to my ear sits somewhere between classic distortion and fuzz. I would still have liked to have had a separate distortion amount control but, even so, the pedal seems just as happy with single-coil pickups and humbuckers. Take a listen to 'Gimme All your Lovin'' by ZZ Top to get a rough idea of how much drive you get! Bringing in the upper octave brightens the sound until at higher settings it becomes more obvious and the overall sound loses some low end.

Turn down your guitar volume and what happens depends very much on the setting of the Balance control. Set this fully anticlockwise and you can clean up the sound as you back off your guitar to a useful extent, although it never gets completely clean. Turn it clockwise and at lower guitar volumes the result is like a fuzz where the transistors have been biased to turn off at low levels, so you are greeted with almost expander-like dynamics and an unnatural, ring-modulator-like harmonic structure (not dissimilar to some of the sounds used by Jimi Hendrix during his Band Of

9V
volume
octave
shadow
sun
balance
amp
input
LIZARD
OCTAVE / DISTORTION
QUEEN
electro-harmonix
JHS PEDALS NYC, USA

Gypsies period). Given the modest cost and the fact that there are only three knobs to play with, the Lizard Queen can conjure up a good range of useful tones covering the requirements of both rich distortion and fuzz. *Paul White*

£ £99 including VAT.
W www.ehx.com

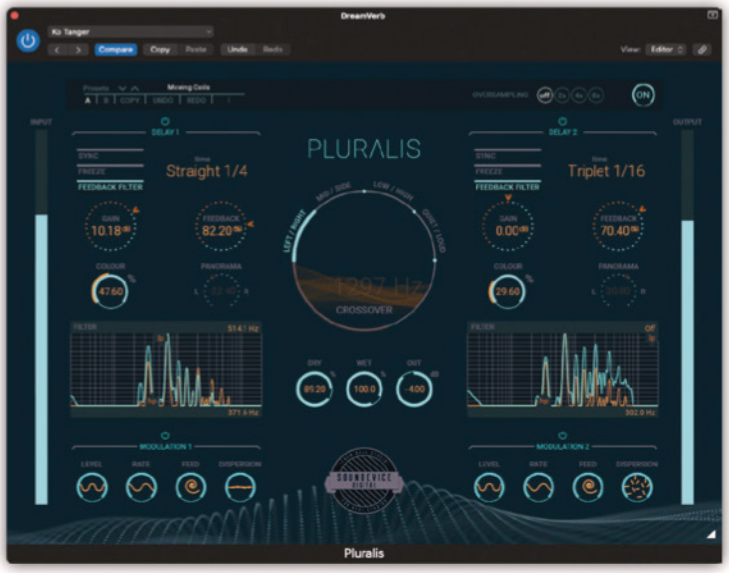
Soundevice Digital Pluralis

Delay Plug-in

Developed by Soundevice Digital and marketed by United Plugins, Pluralis is a fresh twist on the familiar delay — it allows for the creation of effects that are not possible with conventional delays. All the common plug-in platforms for Mac and Windows are supported, and authorisation is via a personalised key file that allows installation on multiple machines.

This plug-in not only offers two delay engines, but also comprehensive filtering and modulation options, and each line can be fed from a 'split' version of the input signal. Splitting can be set by left/right, Mid/Sides, low/high frequency or by level. In Low/High mode, the crossover frequency can be set using the large centre knob, with the various mode settings positioned around its upper edge. In Quiet/Loud mode the centre knob becomes a threshold control.

The delays themselves offer the usual delay time (DAW sync'ed or free), level and feedback controls, but they also have a Colour control, to add some tape-like saturation, and in the last two modes a Pan control too. The graphic area below shows the frequency spectrum of the signal, and the HP/LP sliders can be used to trim


DreamVerb
Compare
Copy
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Undo
Redo
View:
Editor
INPUT
DELAY 1
SYNC
FREEZE
FEEDBACK FILTER
Straight 1/4
GAIN
10.18
FEEDBACK
82.20
COLOUR
47.60
PANORAMA
PLURALIS
MID / SIDE
LOW / HIGH
LEFT / RIGHT
QUIET / LOUD
1297 Hz
CROSSOVER
DELAY 2
Triplet 1/16
0.00
70.40
29.60
OUTPUT
FILTER
DRY
WET
OUT
85.20
100.0
4.00
MODULATION 1
MODULATION 2
LEVEL
RATE
FEED
DISPERSION
SOUNDEVICE
Pluralis

the highs and lows from the signal. Clicking the Feedback Filter bar places this filter in the feedback loop rather than across the delay's input, so that each delay receives progressively more filtering. A Freeze bar allows the current delay tail to be repeated indefinitely. All this processing does, of course, demand a higher-than-average CPU load for a delay. But the results are impressive, and the code has been optimised to keep the CPU demand down to a reasonable level.

At the bottom of each delay section is a row of four controls under the heading Modulation, and these can make a profound difference to the sound. Level and Rate work as expected in adding a pitch modulation but then we find Feedback, which adds a flange-like regeneration to the sound. Apparently this sends an adjustable amount of the phase-shifted signal back into the phase-shifting part of the modulation circuit. Dispersion then adds what sounds like a coarse reverb effect to further ‘shatter’ the delays, which makes it possible to create some unexpectedly dense textures, often adding a crystalline shimmer to the sound but without employing pitch-shifting.

Master controls provide independent level adjustments for the dry and delayed sounds, with a master level control governing the overall output level. Clicking the Soundevice Digital logo takes you to a web page describing the product, and from where the manual can be downloaded.

As expected, there’s a selection of presets to get you started and these demonstrate that it is possible to coax a very wide range of effects from just this handful of controls: plucked guitars and pianos can be turned into crystalline pads that stretch out forever, but there’s also scope for dynamic delays that change with playing volume, or delays that are very different in the right and left channels. In M-S mode, the outer edges of the stereo mix can be treated with different delays to what’s happening in the centre. Splitting by frequency is also musically useful, as it allows one type of delay to be added to the lows and something quite different to the highs. Overall this is an easy-to-use plug-in capable of some musically useful delay coloration that you won’t find elsewhere. *Paul White*

£ Full price £69 including VAT. Discounted to £17 when going to press.
W https://unitedplugins.com

Puremagnetik Tensol

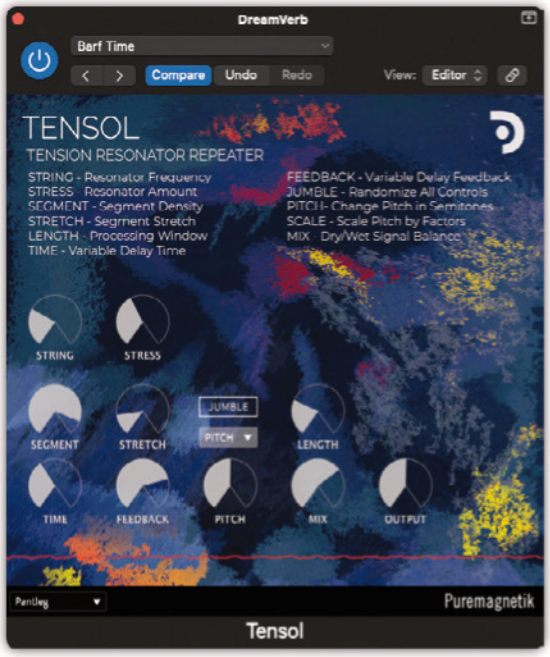
Granular Delay Plug-in

The very affordable Tensol is a curious little effects plug-in that combines granular delay processing with a resonator that’s been modelled on a string under tension. The input runs through the resonator, which has controls for frequency (String) and resonator amount (Stress). After that comes the granular section, which essentially chops the signal into sections which can be stretched and pitch-shifted before being sent to the output. Pitch-shifting can be in musical semitones or a tape-style varispeed. Though there are relatively few controls, there is a Jumble button, which randomises all the control settings... and occasionally this comes up with something worthwhile! Tensol requires an AU, VST2 or VST3 compatible audio host, runs on Mac OS 10.8/Windows 10 x64 and above, and supports Apple Silicon.

> “For a ludicrously low asking price, it offers an easy way to add interesting textured delays to otherwise bland sounds.”

In the granular section, a Segment control adjusts the grain density, while a Stretch control time-stretches the grains. Length determines the size of the processing window while Time adjusts the delay time, but there’s no tempo sync option. Feedback sends some of the delay output back to the input, Mix does the usual wet/dry balance thing, while the Pitch control knob can be switched into scale or pitch modes. In Scale mode the knob controls the pitch-shift as a scaling factor, much like changing a tape speed, while Pitch mode operates musically in semitones or fractions thereof, with a range of ±1 octave.

Despite its apparent simplicity, there’s a fair range of granular-type sounds to be had, and if you’re processing pads or drones then the lack of a tempo-sync function doesn’t really matter — you’d miss it more if processing drum parts or other elements with an important rhythmic function. The input resonator adds a useful timbre to the signal, and most control positions will give you something useful, though I found that I got clicking on occasions after some vigorous adjustment of the controls, despite my CPU meter suggesting that all should be OK. Hitting the Jumble button often comes up with something useful: it’s usually accompanied by the sound of electronic dolphins for a few seconds as the various speed and filter changes come into effect, but you’re still able to create some interesting presets this way.


DreamVerb
Barf Time
Compare
Undo
Redo
View:
Editor
TENSOL
TENSION RESONATOR REPEATER
STRING - Resonator Frequency
STRESS Resonator Amount
SEGMENT - Segment Density
STRETCH - Segment Stretch
LENGTH - Processing Window
TIME - Variable Delay Time
FEEDBACK - Variable Delay Feedback
JUMBLE - Randomize All Controls
PITCH- Change Pitch in Semitones
SCALE - Scale Pitch by Factors
MIX Dry/Wet Signal Balance
STRING
STRESS
SEGMENT
STRETCH
JUMBLE
PITCH
LENGTH
TIME
FEEDBACK
PITCH
MIX
OUTPUT
Puremagnetik
Tensol

Tensol is not going to give you anything like the same depth of processing as top-shelf granular plug-ins such as Arturia’s EFX Fragments, Output’s Portal or Unfiltered Audio’s Silo. But, for a ludicrously low asking price, it offers an easy way to add interesting textured delays to otherwise bland sounds. Because of its lack of tempo sync’ing, you do have to be careful when setting it up to work with rhythmic sounds such as drum loops and the like, but for anything else it’s pretty much a case of ‘anything goes’. You get a few factory presets to get you started, but with the aid of that Jumble button you can build your own preset library in no time. *Paul White*

£ $20.
W https://puremagnetik.com

Ableton Push 3

Controller & Interface For Live

The third generation of Ableton's Push adds audio and MIDI interfacing and the option to go computer-free with battery-powered standalone operation.

SIMON SHERBOURNE

Eight years on since its release, Push 2 didn't seem to be slowing down. It was still the Session controller most Live users aspired to and it enjoyed ongoing updates and integrations with Live's built-in devices. That's not to say that there hasn't been years of speculation and anticipation about how different a next-gen Push might be. And now we know: Push 3 is at the same time remarkably similar, and radically different to what came before.

Ableton have maintained Push 2's form factor and core Live control workflow in Push 3. There are some welcome layout and button rethinks, but the headline new features are things that Push 2 simply couldn't do. The least surprising is the inclusion of onboard I/O: Push is now an audio and MIDI interface as well as a controller. But who imagined that the pad grid, although due for an upgrade, would become a full-on no-kidding MPE controller? Oh, and Push can now be configured as a standalone workstation, running Live internally.

Ableton Push 3
From £879

PROS
- MPE instrument control.
- Clip/scene launching from the screen.
- Built-in audio, MIDI, and host USB connectivity.
- It's got a battery!

CONS
- No Arrangement access.
- No access to clip launch modes and actions.
- Comes with a premium price, and Live Standard/Suite is extra.

SUMMARY

It's a controller, it's an instrument, it's a workstation. I've got this far without saying 'gamechanger', but Push 3 really does set the new standard.

Push provides control of Ableton Live, offering Session view navigation and clip launching, note and drum playing and sequencing, and device/mix control. It can be used in different ways: at its simplest it can be a peripheral to manage a recording session or performance; but it can also be a hybrid physical instrument and drum machine. Workflows that have co-evolved with Push, such as its sequencing and sample chopping, turn the Live/Push combo into a hands-on workstation like MPC or Maschine: a focused hands-on environment for composition and idea capture. It's in this light — Push as workstation or instrument rather than glorified Launchpad — that the new additions make particular sense.

The Hardware

Push 3 looks like Push 2. It's a touch bigger overall, but the pad grid, screen, track encoders and buttons follow the same design. Key visual differentiators are the all-white grid, the replacement of the two leftmost encoders with larger knobs further down, and a jog wheel on the right. I've never liked the rubbery buttons on the Push 2, which sit flush with the surface of the device and feel a bit awkward to me. On Push 3, the buttons are harder and sit slightly proud of the surface. They still require a definite push to register rather than a tap, which can make moving around feel a bit plodding compared to other devices, but they are better. They're also all delineated with raised ridges, which I think will aid accessibility for visually impaired users. The buttons have been reorganised, grouping recording functions to the left, edit functions in a cluster to the right, and so on.

The unit is an absolute tank. It's incredibly solid, with a large metal heatsink on the bottom that doubles as an access panel for the standalone components. Consequently it's heavy: the controller version slightly more so than Push 2, and the standalone nearly a kilo more. None of this is surprising given the additional hardware components.

Along the back panel are quarter-inch headphone and main analogue output jacks, plus another two for inputs. The inputs are switched on screen between line, instrument and 'high' levels, with the latter adding digital gain to accommodate a dynamic mic. An interesting choice is the ADAT I/O. Ableton might have gone for expandability via USB audio interfaces (like MPC and Maschine), but this is actually a really clean and simple option.

Many of us have interfaces and mic pres with ADAT connectivity. I use a Ferrofish line/ADAT converter to run all my synths into my main audio interface. I can just connect one Lightpipe cable to hook up my synth rack directly to the Push, whether for standalone or computer operation. In standalone mode I also enjoyed being able to plug via ADAT into my studio's Pro Tools-based interface and have Live running as a hardware device alongside, sync'ed via Ableton Link over Wi-Fi. Like ReWire reincarnated.

MIDI is catered for via a pair of 3.5mm TRS Type A ports. Then there's a USB host port for directly connecting MIDI keyboards, controllers and synths. A hub can be employed here for up to six devices. I was able to connect my Retrokits MIDI interface to expand the MIDI port count. MIDI and audio device settings can now be managed directly from Push, whether in standalone or controller mode. Devices connected to the USB host port can be used in either mode, although standalone will ignore non-MIDI peripherals like a mouse. Push also has CV outputs as an alternative mode for the foot pedal ports. This requires a TRS splitter to two mini-jacks (not included), the same solution as on Elektron's Analog Four. These appear to Push or your computer as another four audio outputs which you can freely send CV/gate signals to. In Live, or Push standalone, the CV Tools device manages this neatly. Yes, I did try sending audio out of these ports for giggles and it wasn't nice.

Computer connection is now via USB-C, which handles the control connection and computer access to the internal audio and MIDI interfaces.

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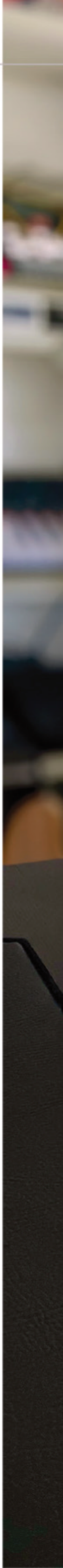
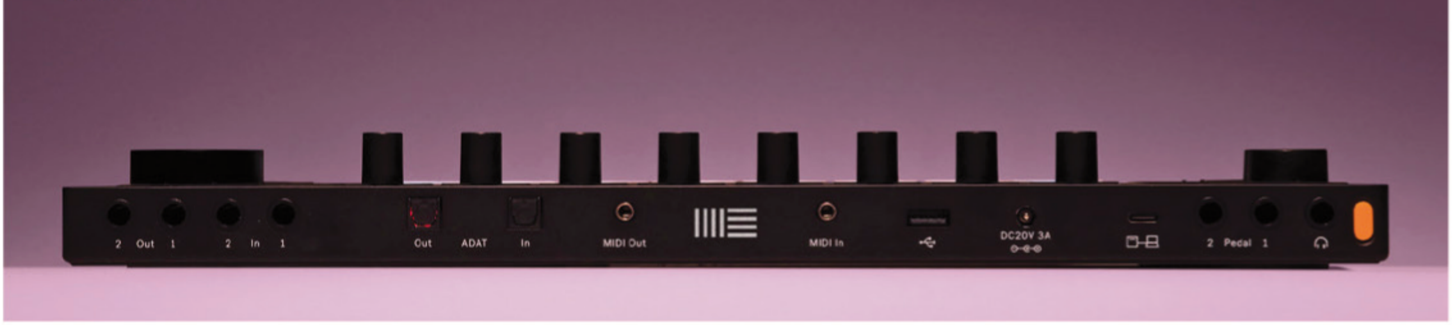




The back panel includes quarter-inch audio I/O, ADAT I/O, 3.5mm MIDI I/O, USB A and C ports, two pedal inputs that can double as CV outputs, and a headphone socket.

» There's no class-compliant direct audio connectivity over USB in the standalone mode. Like Push 2, the controller version of Push 3 can be powered from the computer connection, with the screen dimmed. In the studio you'll want to keep the unusually tidy power block connected. The standalone Push can run off its internal battery, whether in controller or independent modes. This provides about two or so hours of mains-free jamming. Not a lot, but two hours more than any other device bar the MPC Live, and a great backup for gigging. I did discover that you can power and charge the standalone over USB-C if you are using a high-wattage power supply like the one that comes with a MacBook Pro. A direct USB connection to the computer doesn't have enough oomph to keep it charged, although I did think it slowed down its battery use.

New Control Features

Let's look at what's new in terms of controlling Live, all of which is also relevant to the standalone option, which presents the same functionality. While the screen is the same there are new things it can do, some facilitated by the jog wheel. Chief among these is the Session Screen view: a direct representation of the 8x8 clip grid currently focused by the hardware. This feels much more connected to the Session workspace than coloured pads alone, not least because you can see clip names. Turning the jog wheel scrolls you up and down the grid and the view on the pads follows along. The wheel also has a left/right click action so you can navigate horizontally across the session and select tracks. Push-click the wheel and the currently selected scene will launch. Tap the buttons above the screen to launch clips on individual tracks in the focused row. Alternatively you can select and launch single clips by Shift-pressing from the wheel.

Performance

So how much can Push do without the computer? Firstly, in my testing with all the regular kinds of projects I have lying around, the Push 3 didn't miss a beat. A lot of these had VST plug-ins, which I needed to freeze, so I decided to run some tests with the native devices. There's no CPU usage meter on the unit, so I had to just push it until it choked. With the default audio settings, I created a track running the Analog synth device and recorded a clip with an eight note chord playing continuously. I then duplicated the track to see what happened. (There was also a Reverb and Grain Delay plus Saturator on the Returns). Push was able to keep playing at 20 tracks (160 voices of Analog), with things becoming a bit sluggish. By 24 it was beginning to crackle and become unusable. My computer is a MacBook Pro M1 Max with 64GB RAM, and in the same test, using the Push interface with the same settings I maxed out towards 100 tracks. A harsh test as this computer is a beast, but I think it fared well. Also, on the Push standalone you can up the buffer size if you have a big project, and you can also freeze tracks.

You can still nudge or page the main grid using the two cursor clusters. In fact, the screen and pad modes are not joined at the hip. The screen can be running the mixer or device controls while the pads manage the clip grid, just as with Push 2. (My feature request here is that the wheel still scroll the pad grid — it's not doing anything else in this mode). But the huge workflow boost is that with clip/scene launching handled via the screen the pads are freed up to stay in Note/Drum mode. Along similar lines, one of my long-term feature requests is implemented now: you can momentarily switch pad view by holding another mode, then flip back on release.

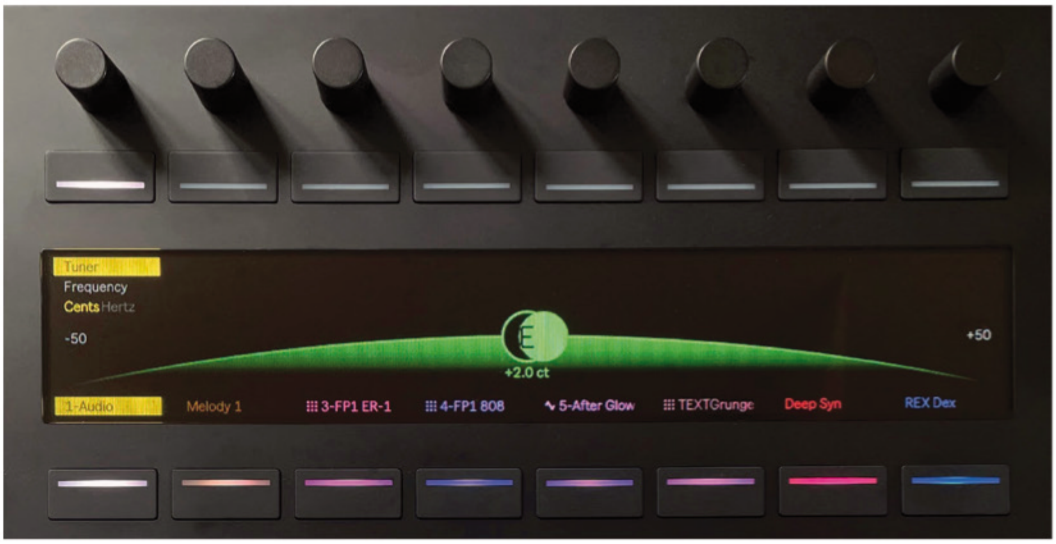
The screen's Clip view now offers a layer of editing function beyond basic clip properties. Previously you could view a piano roll display but only interact with individual notes from the pad sequencer views. In Edit mode the jog wheel is used to select notes, which you can then manipulate with the encoders. You can delete notes, move notes by grid values or fine increments, change pitch, length, velocity, velocity range and probability. The selection method is to scan through notes with the wheel, tapping any you want to select. You can also Select/Deselect All. This works OK, but is not great for selecting a range of notes, or selecting all instances of a note, for example. No doubt this will be fine-tuned in time.

For audio clips there are just two options for editing: you can define a Beat 1 position within the waveform and warp the new range to the grid, enough to tidy up a take or conform a sample or recording to the project tempo. You can't warp within a clip or quantise transients, etc. There's still no visual representation of clip modulations or automation on the Push, which is more of a limitation for standalone use. In Device Control view parameters are still marked as automated with a dot and you see the values change in real time.

Express Yourself

The truth is I've not paid much attention to MPE until recently. My interest was sparked by Ableton's recent efforts to upgrade all Live's internal instruments with MPE capabilities — now we know why they did. MIDI Polyphonic Expression sneaks extra dimensions of per-note performance control down your MIDI port. Push 2 already had polyphonic aftertouch: sensitivity to continuous pressure changes on multiple pads. Push 3's grid is next level, with each pad essentially being an individual X/Y controller. Pads know where they've been hit, how hard you're pushing them, and can sense your finger moving around on their surface.

Nearly all Live's internal instruments can be modulated by these different interactions. Note, velocity and aftertouch were there before, and now we have Note Pitch Bend, Slide and Pressure. You can bend the pitch of a note and add vibrato by moving your finger left and right on a pad. Up/down movements on a pad send Slide modulation, which



The tuner will be a welcome addition to anyone using the Push 3's audio inputs.

is typically used to modulate timbre. Pressure differs subtly from aftertouch and can free you from the constraint of velocity, allowing you to articulate attack and the onset of notes like bowed strings. As well as subtle pitch modulations around a note, you can slide your fingers left or right across multiple pads to glide across larger intervals. The grid interprets this as a continuous bend on the original note and doesn't trigger new notes as you move between pads.

In Push 3's settings you can choose whether notes start exactly in tune regardless of position (and then bend relative to the starting point) or respond to the horizontal position from the outset like a fretless stringed instrument. You can in fact set a tolerance zone in the centre to make it more or less easy to hit perfect notes. Likewise you can tweak the vertical range for Slide expression to suit your style and help avoid running into the next row. You can also turn note bending off altogether, and switch the grid to regular poly or mono aftertouch modes.

To The Max

Push 3 standalone fully supports all the built-in devices in the three versions of Live. It doesn't support VST/AU plug-ins, but if you have Live Suite you can use Max For Live devices. Not everything works at the moment; some devices report that they are missing components they need in the system. There's also no way for the Push to display any custom graphics, which are quite commonly used in Max especially for things like sequencers. However, any Max device that has Push Macros added by its creator (and that doesn't use one of the missing resources) can be operated on the Push standalone. This includes many of the devices included with Live Suite, such as the excellent drum synth modules. My favourite granular sampler worked, but only with presets as you can't currently load samples into the Max devices. Max will be one of the keys to making the Push 3 expandable, so I'm sure more devices will become compatible over time.

Live 11.3 comes with a set of MPE-enhanced patches to experiment with, and most of the Init patches on the built-in instruments respond by default. It's a delight. Vibrato and bend on individual notes in a chord is a particular favourite. And while inevitably slipping into some *Blade Runner*, I found I had the grid constrained to the wrong key: no problem, you can break out and play accidentals at the edges of the pads.

A real showcase of what's possible is the synth drum kit called simply Analog Kit. This is a kit built entirely of sounds programmed in the new Drift synth, making full use of MPE. Each sound responds differently depending on where you hit the pad as well as how hard. If you're quick you can swipe in pitch mods. It's absolutely alive. Note Repeat is lots of fun here, you can easily generate interesting hi-hats and percussion, or clubby snare builds that morph in timbre, pitch and level.

Live Standalone

Despite rumours to this effect, the biggest surprise has to be that Push 3 can be configured as a standalone instrument and workstation. Like MPC and Maschine+, this is powered by an internal processor running a special build of the software behind the scenes. In Push's case this is an 11th-gen Intel i3 with 8GB RAM. Modest, yes, in computer terms, but certainly more powerful than the obvious competition; and remember the Push's processor is dedicated to one thing. Push 3 can be purchased with or without the standalone functionality. Later this year there will also be the option to buy an upgrade kit to add the standalone components to a Push 3 controller. The upgrade doesn't need any special expertise and doesn't affect the three-year warranty.

Out of the box the standalone runs Live Intro, with the various instruments and devices available in that tier of Live (see the box for more on this). If you own Live Standard or Suite, you can log Push into your Wi-Fi and authorise it directly with Ableton. This will lift the 16-track limit and unlock all the built-in devices that come with your software version, and any available sound and instrument packs you own will be available for direct download and installation.

Talking of which, Live Suite comes with over 70GB of soundware, which is no problem for the whopping 256GB SSD inside Push 3 standalone. Unusually there's no SD card option, or even an option to move data over USB. This could be limiting in some situations, but Push 3 has a much better way, which is to use a Wi-Fi link to exchange projects, samples, presets, and so on with Live desktop. Push shows up in the Places section of Live's browser, where you can access the device's library. You can even directly open a Project from the Push, and Live will pull the supporting files locally and open an identical version of the Set. Samples from your Packs will re-link to your local copies. Sets can also be dropped from Live to the Push. When you do this Live suggests you Freeze any tracks with VST/AU plug-ins and use the Collect Files function.

»

Live Version

As a controller Push requires Live on your computer to function in any meaningful way. I did find that it will transmit MIDI (including MPE data) without Live running, but all the lights are out and there's no way to change settings, scales or transposition. Push always comes with Live Intro, which as well as being on Push standalone can be installed on your computer. This gives you the means to control Push (and manage data on the standalone) even if you never plan to use Live desktop.

I'm guessing the majority of those investing in Push will already be Live users, or are planning to become so. If you have the standalone, Live Intro is fairly capable, especially if you're mainly using it for sampling and drum machine duties. It also comes with a couple of extra utility devices: CV Tools and External Instrument, both of which are essential for connecting up with external hardware synths. But it's rather limited in terms of other instruments, so you're really going to want Suite to get all the extra devices and packs to load up that big drive.

» Push in standalone mode is essentially the same experience as operating Live purely from the controller, with the same set of features and limitations. Let's take a moment to look at the workflow for non-Push/Live veterans, then consider how this compares to the full Live desktop experience. A fresh Push project presents you with Session view: each column is a track, with the pad grid representing slots that can contain MIDI or audio clips. Instruments can be added to MIDI tracks, and audio can be recorded or loaded into audio tracks.

Where Push 2 had a Browse button, the new unit has Add and Swap buttons, which both access your collection of instruments, effects and samples. Browser navigation is now aided by the jog wheel. The browser is still a bit basic compared to the graphical, searchable and tagged experiences on MPC and Maschine, and when you go to load a sample you get put into a big list of all audio files on the device (which doesn't happen with Push 2). You can navigate back into the filing system but I'm putting it on the 'room for improvement' list.

As well as the Session layout, Push has a Notes mode where the pad grid transforms into a workspace for playing and sequencing instruments. A selection of note layout schemes are available for melodic instruments or external MIDI devices, and another set for drums. Melodic layouts include a half-and-half notes and step-sequencing mode, or a piano-roll sequencer. For drums you can fill the grid with sounds, but the most common scheme uses the bottom left quadrant for triggers, the top half for sequencing, and the bottom right for selecting different loop ranges within a longer sequence.

As well as the new clip launching view, the screen and encoders can be your mixer or your device control area. Live's devices are nicely integrated with the screen, offering well designed layouts and animated graphics. And as we've already covered, the screen has a Clips view for making adjustments to your clips and their contents.

Recording

With onboard I/O and a standalone mode, one thing that you're likely to be using Push for that feels new is recording. Thanks to the multi-mode inputs you can plug a guitar or bass straight in, and Push provides a tuner device, which makes excellent use of the screen. If you have Live Suite you get the Amp and Cabinet devices to shape your tone, and various multi-effect rack presets.

The other main workflow I tested was connecting a synth to the inputs for monitoring and recording, both sequencing from Push itself, or clocking devices with their own pattern playback. Like Live software, the standalone uses the External Instrument plug-in to manage the routing of outgoing MIDI and incoming audio. This also gives you a timing offset to mitigate latency. With the default settings latency in this scenario was minimal, but you can squeeze it down further by dropping the audio buffer down to 64 or 32 samples.

MIDI (and CV) control of external synths benefits from the MPE pads where supported, but it's not possible to control parameters via MIDI CC. In Live desktop, you need to use a suitable Max device for this, and the one I have didn't work on Push. What does work in Push standalone are Freeze and Flatten. If you've got an external synth being sequenced from Push, you can Freeze clips and they will be automatically captured as audio. Flatten goes another step and turns the track into an audio track, a fantastically neat workflow.

Missing Links

While it's remarkable that Push standalone has feature parity with Push as a controller, it does mean there are gaps. Anything that you would reach for the computer to do in controller mode is, well, out of reach when running standalone. The obvious case is access to the Arrangement. Like the Note app, Push is focused on Session workflow, with no linear tracks or inherent pattern chaining/song mode features. I was really hoping that Push standalone could work as a multitrack recorder for DAWless scenarios, or be able to provide a semi-automated structure for a live performance.

Of course you can make recordings as long as you like, recording into single clips. But consider a typical workflow: you create a beat in Push using a series of scenes for different song sections. Now you want to record a vocal performance alongside this. If you try to record audio, as soon as you change scene the audio track will drop out of record. In Live desktop you could record the vocal into an Arrangement track, but even staying in the Session you'd have options. You could decouple the audio track from scene changes (remove clip stop) or you could enable the Start Recording on Scene Launch setting, which results in a continuous record pass split across scenes. Neither of these options is available in standalone.

There's also no access to clip launch modes, follow actions, launch quantisation, etc. Follow actions, which let you choose an action that happens when a clip ends, would let you create chains. Push will in fact obey any of these settings for sets that you transfer from your computer. This gives you a way to build a live performance: pre-building an arrangement from chained clips, and including single-shot clips and legato unquantised clips for breakbeats and fills.

Interestingly, Live's Arrangement is not absent in the Push, it's just not accessible. This leaves open a hack for laying down an arrangement. The feature I most love in Live is that you can put its Arrangement (linear tracks workspace) into record, then perform a series of clip and scene launches in the Session, and all of this gets

Undo
Save
1-808 Core
2 Drift
3-Drift
4-Wavetable
5 Vocal Audio
6 Guitar Audio
7 Drum Machin
8 Perc Audio
M
S
Tap Tempo
Quantize
Fixed Length
Automate
New
Scale
Layout
Repeat
Accent
Double Loop
Duplicate
Convert
Delete
Octave
Octave
Shift
Select

Push 3 has gained a little size on its predecessor, at 380 x 318 x 29mm versus the 378 x 304 x 26mm of Push 2. Not bad considering it now contains an audio/MIDI interface and (potentially) a computer. Where it's gained a lot, though, is in weight, up to 3.95kg from Push 2's 2.71kg.

captured into the tracks. You can in fact still do this in Push standalone by Shift-pressing the Record button. This is basically a one-time deal as there's no way to then play back what you did, but if you save and transfer the set to your computer your arrangement will be there.

Conclusion

Push 3 stands apart from its peers through sheer elegance of design and cohesiveness of workflow. It may seem a minor thing but the ease with which you can move material back and forth between computer and hardware makes all the difference — the kind of difference between whether you use a bit of kit or not. It feels like a unified instrument rather than a science project. There are some gaps in the standalone functionality, things that were simple enough to achieve on the computer but now need hardware access. And as Ableton look at the ongoing development I really hope they implement Arrangement (as Akai eventually did on Force), or at least some way to combine linear recording alongside scene-based song structure. Push's primary focus has always been idea capture and jamming but I think it's reasonable to expect arrangement and finishing to be within the scope of a premium standalone. Then Push really will be the one device to rule them all that I seem to always be chasing. Anyway, transcending comparisons with the other standalone workstations is the MPE pad grid, and its interaction with the built-in devices. I've gone from 'meh' about MPE to evangelical in the space of a couple of weeks. Regular old keyboards suddenly feel a bit stiff and lifeless; there's no going back. Ableton have always framed Push as an instrument rather than a controller, and Push 3 realises that vision. ■

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How I Got THAT SOUND

George Seara • Shawn Mendes 'Treat You Better'

JOE MATERA

Juno-winning and Grammy-nominated Canadian mixer and recording engineer George Seara has worked on numerous multi-platinum albums since he began his career back in the mid-1990s, with credits including Shawn Mendes, James Bay, Demi Lovato, Alessia Cara, Rihanna, Drake, Herbie Hancock and Sting, to name but a few. Asked to pick a favourite sound from his catalogue, he chooses the vocal on 'Treat You Better' by Shawn Mendes.

"I get asked about this song a lot, and how I mixed it. The song was mixed by a few others as well, and my mix was ultimately chosen as the version that everyone preferred. I'm just humbled to be a part of it all. Shawn is the real deal, a once-in-a-generation talent, and I'm always inspired working with him. His dedication and passion for his craft is noteworthy and I think he really cares deeply about what he's writing and how he's singing it, and that it's authentic.

"I mixed this single on an SSL 6000E console with VCA automation. The console is punchy and has an edge when pushed, which I like. Mixing analogue is how I learned. It's very fast for me, recalls are a snap, and I just love having physical faders and EQs to work with. I can use a gentle touch or push the mix bus for a more aggressive sound, where fitting, and I like that it's pliable and can provide sonic 'colour'."

Up Close & Personal

"'Treat You Better' is a moody, introspective anthem, and Shawn's entry vocals draw you in close right away. Shawn's lead vocals return on the console, which had an Avalon 2055 equaliser, Empirical Labs Distressor [*compressor*], and a Retro Instruments 176 Tube Limiting Amplifier on the insert. My aim was for the vocals to be warm, and present, with close proximity at the entry. I started with a 2:1 desk compression ratio with a fast release and +3dB at 100Hz and +3dB at 16kHz desk EQ, with a high-pass at 70Hz. The Avalon 2055 EQ had a mild presence boost above 15kHz while the Distressor was set to 4:1, with medium attack and a fast release. The Retro 176 was also set to 4:1 medium attack and fast release, and side-chain filtered at around 70Hz.

"I absolutely love these compressors on vocals! In addition, I often use a TC Electronic System 6000 for reverb and effects, and I used a Large Warm Hall preset, as well as a Pitched Stereoizer preset for 'widening' very slightly. The Large Warm Hall sound is magnificent and it's one of my favourites for adding depth and space, while retaining intimacy.

"With pop music, I like to work on vocals early on in the mix, so that they are at the forefront, and often begin the mix by getting a nice quick balance of key instrumentation and then move to vocals. From there I bring in other instrumentation around the vocals. In this case, I started with entry guitars and vocals. I will no doubt spend equal time on making sure that the drums, the foundation, feel great and hit hard, along with the rhythm section throughout." ■■■

Hear The Sound

W https://open.spotify.com/track/3QGsuHI8jO1Rx4JWLUh9jd

W www.youtube.com/watch?v=lY2yjAdbvdQ

SHAWN MENDES TREAT YOU BETTER

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Solid State Logic UF1

DAW & Plug-in Controller

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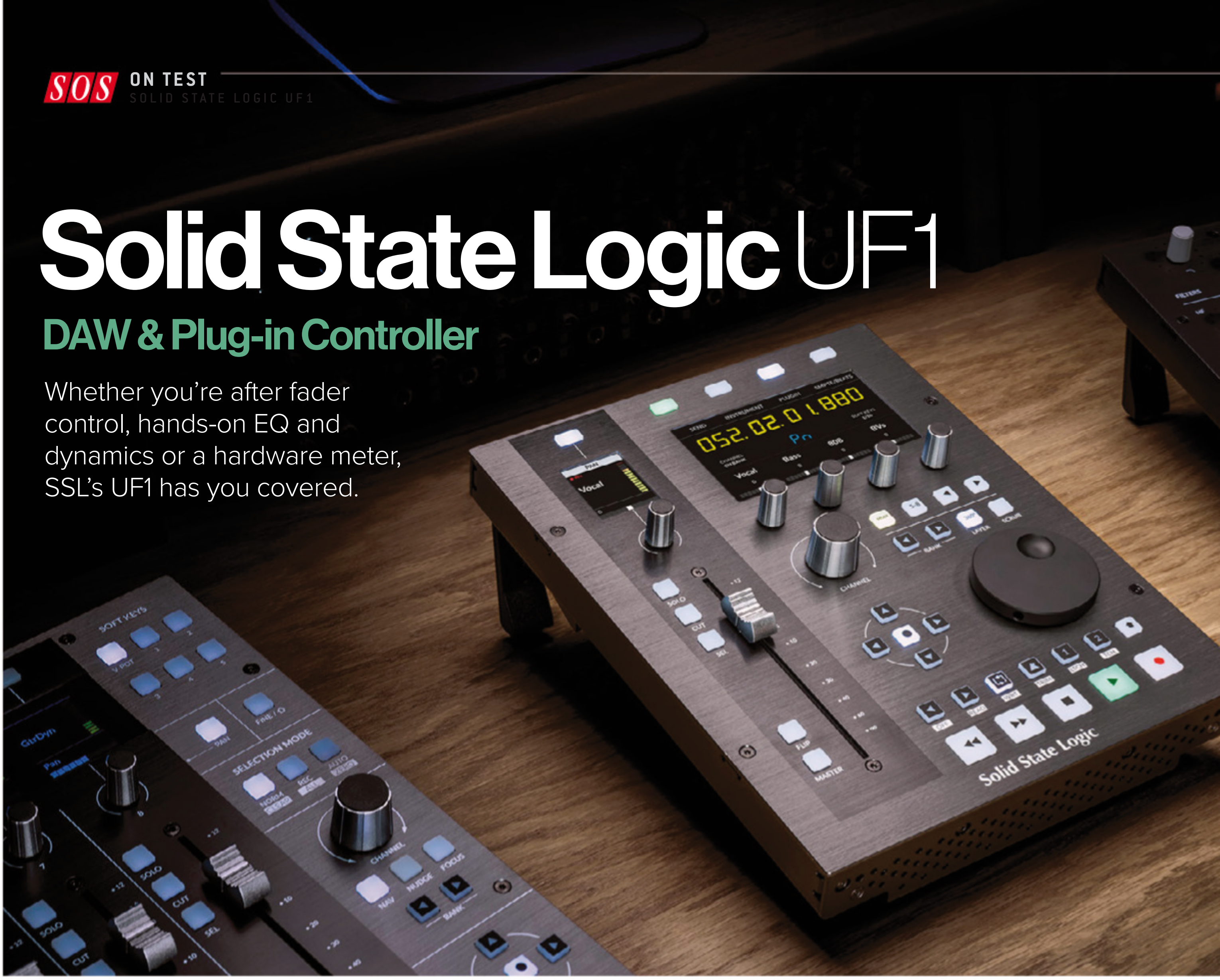


SAM INGLIS

Sooner or later, most DAW users get frustrated with mice. Software mixers still take their design cues mainly from hardware consoles, and it's natural to want faders and buttons under your hands as well as on screen.

However, fader-based control surfaces can themselves be a source of frustration. The now-ancient MCU and HUI control protocols are bottlenecks, limiting the type and amount of information that can be transmitted between DAW and controller. And although EUCON offers the potential for deeper and richer integration, it's a proprietary Avid format and can't be exploited by other controller manufacturers.

The older MIDI-based standards still work fine for basic fader and transport control, along with related tasks like writing automation, navigating through projects with a jog/shuttle wheel and scrubbing audio. But plug-ins are now central to mixing, and it's pretty hard to create an MCU or HUI surface that can offer effective control over all of the countless different compressors, reverbs, EQs and so on that are on the market.

More Than MIDI

For this reason, some manufacturers have begun to bypass the MIDI-based control protocols built into DAWs. Softube's Console 1, for example, communicates not with the DAW itself but with Console 1 plug-ins on DAW channels, via a software application running in the background. The great benefit of this is that you get perfect mapping of hardware controls onto on-screen EQ and compressor parameters. The drawback is that it's another proprietary system and won't work with your choice of third-party plug-ins.

Since Console 1 was launched, Softube have been able to work with software developers to improve DAW integration. Most DAWs are now able to pass track name information to Console 1, for example, while the more recent Console 1 Fader surface can adjust channel volume and pan directly within the DAW mixer, where supported. What started as a closed,

Solid State Logic UF1

£599

PROS

- Excellent build quality, with smooth fader, high-quality colour screens and a very nice jog/shuttle wheel.
- Provides comprehensive transport and session navigation facilities alongside detailed single-fader channel control.
- The main display can be turned into a very useful hardware meter.
- Offers handy additional plug-in control when combined with SSL's UC1.

CONS

- It's not SSL's fault, but their units can't offer the level of control over third-party plug-ins that is possible with EUCON.
- SSL's Plug-in Mixer is not quite the finished article yet.
- Wall-wart PSU with too-short cable.

SUMMARY

Solid State Logic complete their trio of controllers with a neat device that offers much more than just transport, navigation and single-fader channel control. The UF1 is also a highly specified hardware meter, a plug-in controller and more!

proprietary system is slowly becoming more of a team effort, but Console 1 is still very much its own thing and doesn't use HUI or MCU.

Solid State Logic are another manufacturer who have developed proprietary software to allow their controllers to do things not possible with MCU and HUI. Unlike Softube's OSD, though, SSL's 360 is intended to augment rather than replace the older MIDI-based standards.

The SSL UF8 is thus a more or less conventional eight-fader MCU and HUI controller, but with added powers. The Layers system in SSL 360 allows it to be tailored to your DAW of choice, and permits its soft keys to be used to trigger macros: sequences of keyboard shortcuts that might not necessarily be assignable to single keystrokes within the DAW itself. The UF8 was reviewed in *SOS* March 2021: www.soundonsound.com/reviews/ssl-uf8.

By contrast, SSL's UC1 resembles Console 1, in that it's primarily designed to control proprietary plug-ins rather than the host DAW. Its control layout closely mirrors the user interface of SSL's 4K B and Channel Strip 2, which emulate the preamp, EQ and dynamics from two different eras of SSL console. It even has a separate section for controlling bus compressors, and can be paired with another SSL plug-in called Bus Compressor 2 for this purpose.

The introduction of the UC1 also brought a new element to the 360 software: the SSL Plug-in Mixer, which provides a console-style representation of the 4K B, Channel Strip 2 and Bus Compressor plug-ins currently active in your DAW. (Buying any SSL controller gives you a six-month licence for all SSL plug-ins, and UC1 buyers get perpetual licences for these three.)

Flying Solo

The new UF1 is the third controller in SSL's range, and occupies the middle ground between the UF8 and the UC1: it's both a DAW controller and a plug-in/Plug-in Mixer controller. Like the others, it's beautifully made, with a solid metal chassis, high-quality controls and bright, clear, high-resolution colour displays that are clearly visible from any angle. Optional legs bolt on to the underside, allowing the UF1 to be set up at a wide range of angles relative to your desktop. And, of course, it sits perfectly in a row with one or more UF8s and/or a UC1.

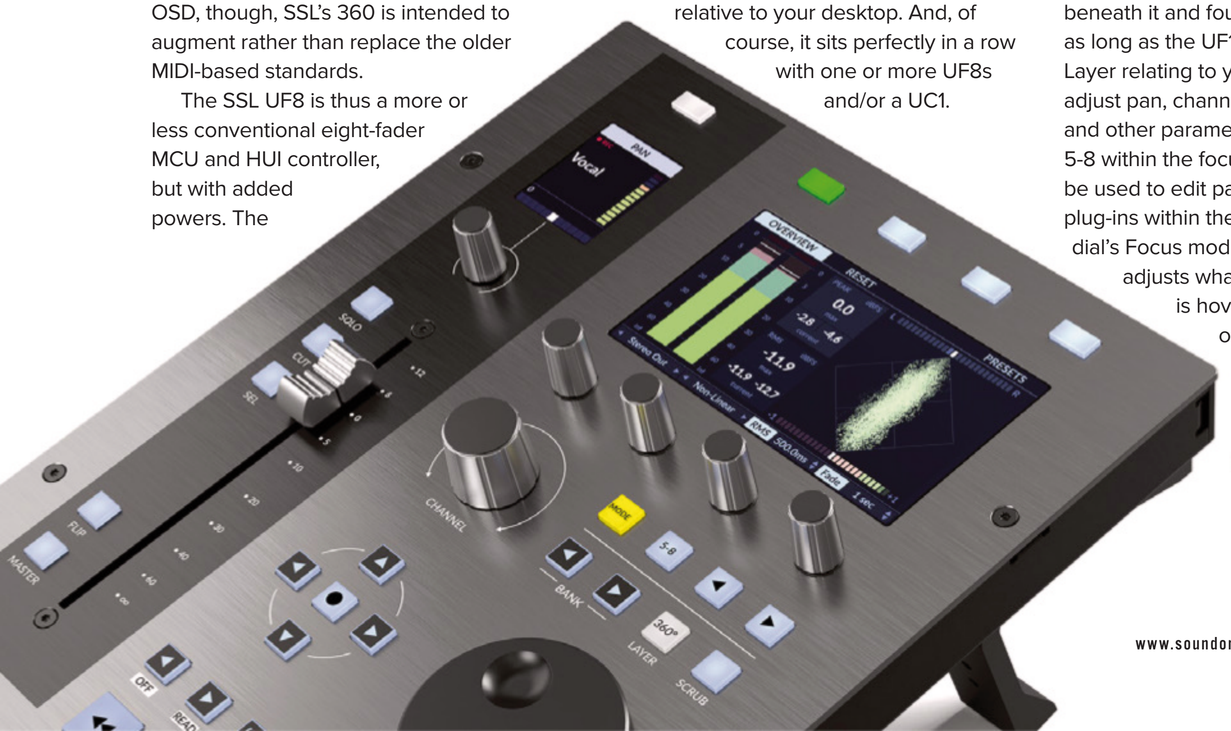
The UF1 talks to your computer over USB, and has a Thru socket that allows multiple SSL controllers to be chained to one computer USB port. However, whereas the UF8 shipped with a laptop-style 'line lump' power supply, the UF1 comes with a wall-wart, and if you have the UF1 positioned at the front of a desk, you may well find the cable isn't long enough to reach floor- or wall-mounted power sockets behind the desk.

One Over The Eight

Some of the UF1's functionality overlaps that of the UF8 and UC1, but in most respects, it's designed to be complementary to both. The left part of the panel is basically a single channel from the UF8, with the same high-quality, long-throw motorised fader, small display and rotary encoder. Depending on your mixing philosophy, this may be all you need: many people find it natural to record automation passes one channel at a time, and don't engage with the old-school approach of moving multiple faders simultaneously. As an alternative, there's also a Master button that switches the fader to the DAW master channel; this is supported in MCU-compatible DAWs, but not Pro Tools.

Although it has just the one fader, the UF1 addresses a bank of eight channels within the DAW. As on the UF8, the primary function of the Channel encoder is to shift the focus left or right along the DAW mixer, and there are also Bank arrow buttons that move in steps of eight channels. By default, the 'focus' channel addressed by the fader and its associated display is the leftmost of the bank of eight, but this can be changed, and some control is available over other channels too, courtesy of the large display at the top right. This has four encoders beneath it and four buttons above it, and as long as the UF1 is operating in the Layer relating to your DAW, these can adjust pan, channel volume, send level and other parameters for channels 1-4 or 5-8 within the focus bank. They can also be used to edit parameters on third-party plug-ins within the DAW, but the Channel dial's Focus mode, whereby the encoder adjusts whatever control the mouse is hovering over on screen, is often a more elegant way of doing this.

»



🟩 The UF1's large display can play host to several different types of meter.

»

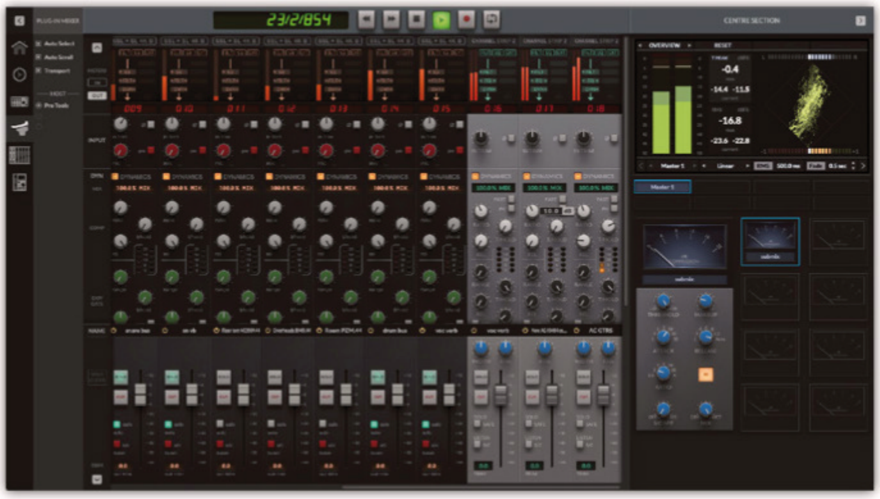
The Channel encoder is joined by the same cross-shaped group of arrow keys that appear at the lower right of the UF8, and they perform the same roles here. These roles are DAW-dependent, and in Pro Tools, for example, the arrow buttons can either move the edit cursor around or change horizontal and vertical zoom level.

The UF8 has assignable soft keys that can be used to provide transport control, but they aren't labelled or ideally positioned for this job. This is one of the gaps that the UF1 fills in in style, with a row of five large, illuminated buttons placed exactly where you want them. Above this is another row of six buttons that default to oft-used secondary functions such as moving to the start or end of the project, enabling the metronome, switching on loop recording and so on. A Shift button turns these into a comprehensive set of automation mode selectors for the focus channel, but it's also possible to select alternate functions within 360. For example, I think most people will find the ability to jump back and forth between markers more useful than the ability to jump to the start and end of the session, and that's supported as an alternative.

Finally, the largest control on the UF1 is a large and very nicely weighted jog/shuttle wheel. It jogs, it shuttles, and it also scrubs, courtesy of the button immediately above.

Outside The Box

Considered as a single-fader MCU/HUI controller, the UF1 delivers a lot of attention to detail, but the nature of those protocols means it pretty much has to work a familiar way. So, although it's more capable than cheaper alternatives like the PreSonus Faderport, it's not fundamentally different. The built-in displays, proper jog/shuttle wheel and secondary control over four channels certainly elevate it above those devices, and given that the UF1 is a lot more expensive, anything less would be a disappointment. What is new and unfamiliar, at least to me, is the Plug-in Mixer and the capabilities the UF1 either shares with the UC1 or adds to it.


Each 4K B and Channel Strip 2 plug-in within your DAW has a corresponding channel strip in the 360 Plug-in Mixer. Up to eight SSL Meter and Bus Compressor 2 plug-ins can also be accommodated.

When you engage the Plug-in Mixer Layer, the Channel encoder now steps through instances of SSL plug-ins rather than DAW mixer channels. Since each instance of a 360-enabled channel strip has its own Plug-in Mixer channel, that also means it's stepping through the channels within the Plug-in Mixer. DAW channels with no plug-in instantiated don't appear in the Plug-in Mixer, and conversely, a single DAW channel can be represented two or more times if you use multiple instances of 4K B or Channel Strip 2.

Default channel ordering within the Plug-in Mixer depends on your DAW; with Pro Tools, channels initially appear in the order in which the respective plug-ins were added to the session. However, you can click in the channel name field and drag left and right to reorder. As with Console 1, channel names are transferred automatically from the DAW.

The channel strip plug-ins have their own faders, solo, mute and pan controls. Using these rather than the equivalents in the DAW mixer means there's less need to constantly switch back and forth between the DAW Layer and the Plug-in Mixer Layer, but you'll need to do some careful setup at the start. For example, if you have tracks feeding aux busses in your DAW, you'll need to make sure that solo safe arrangements are in place within the Plug-in Mixer. (The version I tested had a minor bug with Pro Tools only, whereby soloed signals only appear in the left channel; SSL are aware of this and it will be fixed soon.) Similarly, the plug-in pan controls only do anything in plug-in instances that have a stereo output path.

Parallel Lines

The channel strip plug-ins are pretty deep and powerful affairs, with filters, four-band equalisers, compressor and expander/gate, and various other features. These all fit comfortably within a single plug-in window in your DAW, but when arranged vertically within the Plug-in Mixer, will likely prove too long for most monitors. You can choose to hide the fader section at the bottom and the routing pane at the top, but even so, some scrolling is needed. As it stands, the Auto Scroll option scrolls the mixer horizontally to make sure the focus channel is in view, but won't automatically scroll up and down if, for instance, you happen to be adjusting an EQ control that's disappeared off the top of the screen. The Plug-in Mixer is a work in progress, though, so this may well be on the map for a future version.

For obvious reasons, the UC1 is designed to match the square layout of the channel strip and bus compressor as they appear in the DAW plug-ins, rather than the long, thin arrangement they have as Plug-in Mixer channels. Once you enable the Plug-in Mixer Layer for UF1, it too can control these plug-ins, but that visual correspondence is missing. In this mode, the UF1 makes up to four plug-in parameters at a time adjustable using the rotary encoders below the screen. These are sensibly grouped such that, for example, all the parameters for a given EQ band are shown together, and the soft keys at the top are used for relevant buttons and switches such as EQ in/out. The large screen also provides a graphical representation of the EQ settings, while dynamics settings are shown within the smaller display above the fader.

All of this works well, but as far as plug-in control is concerned, the UF1 is intended more as a useful extension to the UC1. This is brought home by the order in which the 4K B plug-in parameters appear. The first page collects together stereo Width, Mic preamp gain, Out Trim and Compressor Mix — which have nothing in common except that they lack dedicated controls on the UC1. You wouldn't buy the UF1 just for plug-in control, then, but it would be an excellent partner to the UC1.

Look Sharp

However, that doesn't mean the UF1 is doomed to play second fiddle to the UC1 within the SSL 360 ecosystem. It has its own starring role to play, courtesy of its large colour screen. This makes possible a sophisticated metering mode, which can be accessed both from your DAW Layer and from the Plug-in Mixer Layer.

To turn the UF1 into a meter, you first need to install the SSL Meter plug-in (UF1 buyers get a perpetual licence for this, as UC1 owners do for the channel strips and bus compressor). You can then instantiate up to eight instances in your DAW session, and hitting the UF1's Mode button a couple of times will bring up one of the plug-in's three meter panes within the UF1's display. These are, respectively, an 'analogue' view which shows a pair of VU or PPM meters, a 32-band real-time spectral analyser, and an Overview pane with various different numeric readouts and a goniometer. All of these are highly configurable if, for example, you need to calibrate the VU meters with respect to a specific output level from your audio interface. The only obvious thing missing at present is LUFS loudness metering, but SSL hope to implement this soon.

I think the metering will be a major selling point for the UF1. Good hardware meters can be very expensive, and the convenience of having this feature integrated within a DAW controller is hard to overstate. Plus, of course, switching the main display to Meter mode doesn't affect the UF1's capabilities as a fader or transport controller.

Points Of Detail

Thoughout this review I've drawn comparisons between SSL's controllers and those of other manufacturers. Operationally, the UF8-UC1-UF1 ecosystem resembles Softube's Console 1 and Console 1 Fader, but its most obvious rival is probably Avid's proprietary EUCON control scheme. This system allows Avid's controllers to combine HUI-style fader control with detailed editing for any plug-in, without the need for a separate software framework like the Plug-in Mixer. Within this scheme, the UF1/UC1 combo faces off against Avid's Dock plus your choice of tablet running Avid Control.

The fact that Avid controllers have access to EUCON gives them a head start in terms of functionality, and I'm sure SSL would prefer not to have had to develop their own proprietary plug-in control system to work around the limitations of MCU and HUI. But considered as a fader and transport controller, the UF1 wants for nothing — and when it comes to plug-in control, SSL's system makes up in immediacy what it lacks in universality. Control over third-party plug-ins is limited, but control over SSL's own plug-ins is absolute, and as Console 1 proved, there's something about a perfect correspondence between controller and plug-in that is lost in any generic system, no matter how flexible it is. Finally, I can't finish this review without mentioning quite how good these SSL plug-ins sound. The 4K B channel strip, in particular, really does deliver that up-front, in-your-face quality that makes SSL consoles so enduringly popular.

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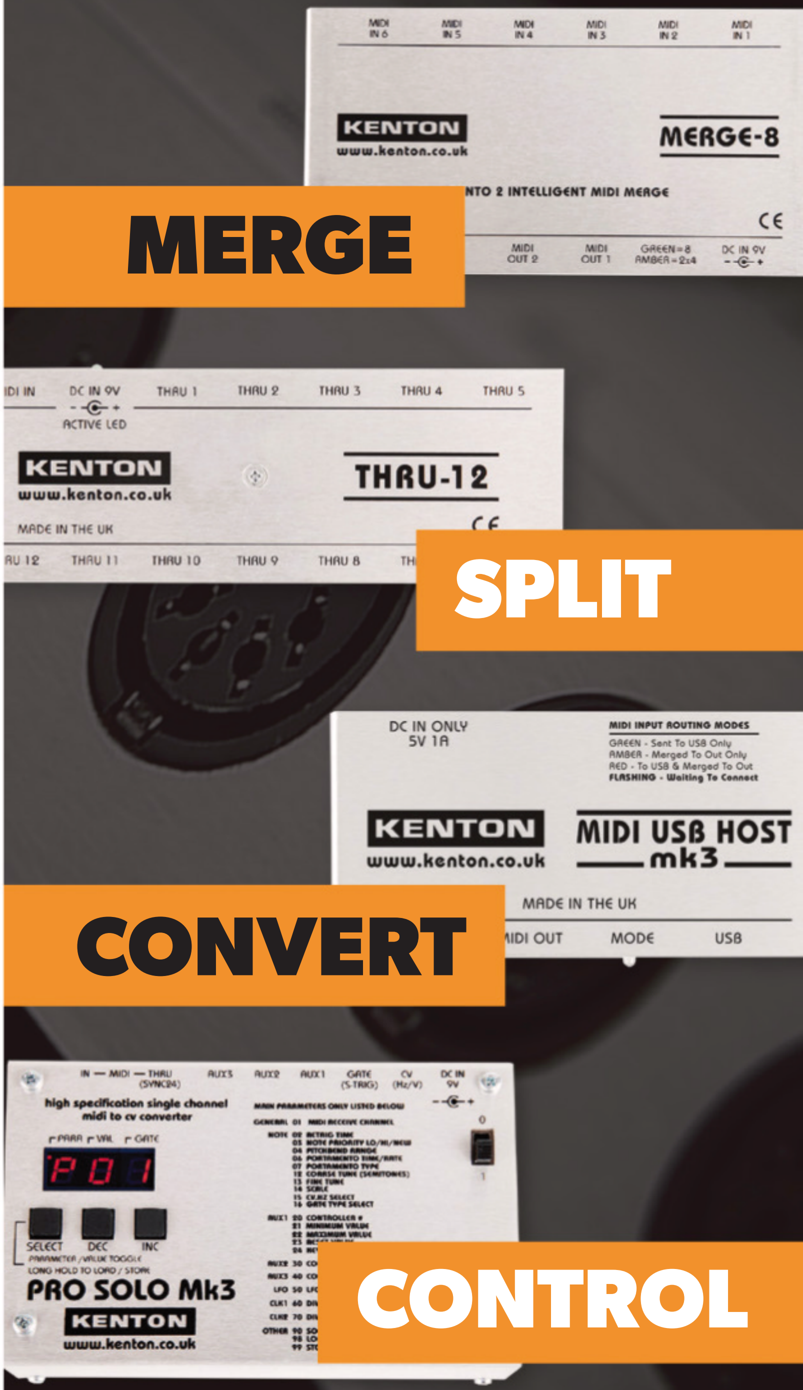

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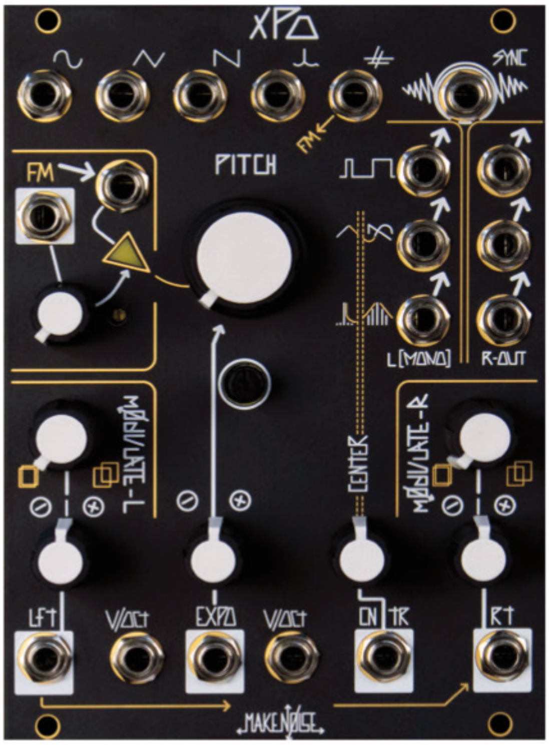
MODULAR

Make Noise XPO

Eurorack Module

Make Noise need almost no introduction. Among the prime movers in the Eurorack explosion, former Moog employee Tony Rolando and his crew broke on to the scene with a distinctive street-art-meets-hieroglyphics aesthetic and a certifiably maverick attitude towards circuit building.

On first glance you might think their new flagship standalone oscillator, the XPO, is simply a sequel to the much lauded DPO, and although it does incorporate elements from that design, the abbreviations indicate otherwise. While the DPO is a Dual Primary Oscillator, the XPO is a Stereo Prismatic Oscillator, departing from the DPO's architecture in several key areas. Rolando, interestingly, did purportedly refer on occasion to the DPO as a Dual Prismatic Oscillator, indicating that the journey to the XPO has been a long one. The genesis point of the XPO, according to Rolando, can actually be traced more accurately to the voice panning of the Oberheim OB-8. “So simple, so effective,” he wrote on the Make Noise website. “Then it occurred to me, why couldn't PWM [*pulse-width modulation*] be stereo? It would be very effective at creating a stereo image since the amplitude of both left and right would be almost identical, while still offering dramatic stereo timbral shifts.
I thought to myself, surely somebody has done stereo PWM already.
I needed to research this idea. So I wrote about it in the notepad I keep on the nightstand and I eventually fell back asleep.”



In his extensive research following that dreamlike revelation, Rolando came to realise there was in fact a dearth of stereo applications of PWM, and I suppose the rest is history. A monophonic all-analogue oscillator based on a triangle wave core, the XPO offers three sets of stereo outputs, five mono outputs (that's a whopping 11 separate outputs), two summed volt-per-octave inputs and many, many modulation possibilities. There are elements of Make Noise's aforementioned DPO and QPAS filter here, not to mention plenty of the trans-American synthesis school-mixing characteristics that (literally) made the name of the semi-modular 0-Coast. The XPO is a beastly thing with legion tricks up its sleeve, that I daresay I would not exhaust with weeks of experimentation.

In a world where spatial audio is marketing itself as the sound of the inevitable march of progression, it's easy to forget that there is still infinite uncharted territory and abundant creative potential when it comes to our old friends, L and R. Make Noise, it would appear, are invested deeply in this idea. There's the now-iconic, two-channel Maths function generator, the aforementioned DPO dual VCO, the stereo Mimeophon ‘audio repeater’ and of course the QPAS stereo filter, with which the XPO is alleged to pair famously. Make no mistake: stereo is as lush a sonic territory to explore as it has ever been, and then some.

As mentioned, the XPO has an astonishing 11 possible outputs. These consist of a top row of singular sine, triangle, saw, spike and sub-octave outputs, alongside stereo pairs of PWM square-wave outputs, sine-triangle variable ‘Vari-Timbre’ outputs and sine-based wavefolder outputs. Of these, we might chalk the triangle, saw and square-wave outputs up as East Coast circuits and the spike, sub, Vari-Timbre and wavefolder outputs as West Coast (perhaps even along with the FM-friendly sine wave). More specifically, the spike wave nods to the Buchla Music Easel synth, while the Vari-Timbre and wavefolder pay homage to the Buchla 158 and 258, and the 259, respectively. Remember all that about love letters? This is what I mean, and isn't it fantastic?

In any case, immediately upon powering up my XPO I was able to populate an entire mixer with glorious, wide washes of oscillations, processing and filtering any combination of them through various other modules to create a truly humongous, multifaceted drone without so much as patching anything into the module at all. While the XPO's mono outputs can take any comparable Eurorack oscillator to task, here they are rendered relatively utility, since the XPO's focus is undoubtedly on its three sets of stereo outputs. Two Modulate knobs shape the output of the left and right sides independently of one another, eliciting a different response from each stereo set: they modulate the PWM square-wave outputs' pulse width (in case that wasn't clear already), they graduate the Vari-Timbre outputs from sine to saw, and on the wavefolder they, well, fold the wave. For the latter two sets there's an additional control, Center, which ostensibly sets the core wave shape that ‘feeds’ each of these two circuits when their respective Modulate controls are turned up sufficiently.

It doesn't stop there with the XPO's modulation potential. Two FM inputs — one linear and one exponential — offer more sound sculpting and self-patching potential. This also helps to explain why there are so many different outputs on the XPO: it's a veritable powerhouse of harmonics in almost every way you can think of. The linear FM is normalled to the XPO's sub-octave oscillator, which is a nice touch. This means that even when unpatched, the linear FM attenuator can achieve anything from subtle growl and quasi-distortion to tonally unhinged chaos. There is also a rather unique-sounding oscillator sync input that doesn't quite adhere to the laws of soft or hard sync; instead it creates more of a 'rippling' effect (to use the apt term put forth in the XPO's manual) that once again endows the XPO's sonic signature with fascinating, at times guttural timbres.

Two volt-per-octave inputs complete the XPO panel, but don't be fooled into thinking this indicates duophony. These might look like they promise to control each side independently, something not helped by their placement on the panel, but in reality are identical if used individually and become summed when used together. This isn't so much a criticism of functionality as it is just a bit of a red herring, since duophony would in many ways undermine the whole point of the XPO: which is to explode its waveforms into as many harmonic splinters as possible while retaining the integrity of that big, brash, gloriously bold singularity that is a monophonic analogue voltage-controlled oscillator. It may feel like spatial audio is the talk of the town in 2023, but stereo's horizon is still expanding and the XPO ably demonstrates that. It's wide and lush, replete with complexity and texture, and a worthy addition to the Make Noise family. Marvellous. *William Stokes*

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OXI Instruments Coral

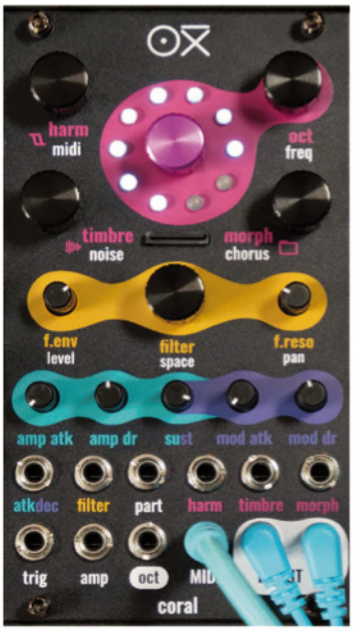
Eurorack Module

The Coral from OXI Instruments is a polyphonic synthesizer in a compact Eurorack module. It takes algorithms from the Mutable Instruments Plaits firmware, expands them out across eight voices and builds in a filter and VCA to take it from oscillator to synthesizer. We've seen Plaits sound engines expressed in modules before and polyphonically in synthesizers like the Arturia MiniFreak. However, what the Coral does differently is that it opens it up to eight parts of multitimbrality.

Coral is an eight-voice, eight-part, 10-sound-engine synthesizer in a 14HP module. The sound engines include virtual analogue, waveshaping, FM, wavetable, supersaw, and string. Then there are three percussive synths covering hi-hat, snare and kick drum. And finally, in a departure from the Plaits, we have a wave player sample engine. All the sound engines get three parameters to play with for shaping, detuning, folding, pulse widthing, indexing and feedback depending on the chosen algorithm.

The middle section of the module houses a resonant low-pass filter for tonal shaping. Beneath that is a slightly odd pair of envelopes. One side handles the VCA while the other plays with the filter. They have a shared Sustain control, and the Decay knob is also the Release knob. It makes for simple enveloping but sometimes feels like it is not always working with you. Beneath that, you have a good range of CV inputs for modulation and CV control. Unusually for a Eurorack module, the Coral is designed primarily for use over MIDI, the assumption being that you'll plug in a MIDI controller.

The Coral uses a ring of LEDs around the central purple/pink encoder to navigate the sound engines. To reach deeper levels or secondary parameters, you must hold in the encoder while rotating it or moving other knobs. As a system, it works well enough and avoids the need for a display or menu buttons. However, your hand is often trapped there holding in the encoder and can get in the way of the LEDs. It's at its most clumsy when allocating parts to MIDI channels, with lots of holding, turning and trying to see around your hand, but it's an annoyance rather than a problem.



You can share the eight voices between anything up to eight parts by giving them a different MIDI channel. So you could allocate the kick, snare and hi-hat engines to three channels, use the virtual analogue engine for a monophonic bass line on another channel, and you are left with four voices for a multi-oscillator supersaw pad on channel 5. You have level and pan control over the individual parts and a Space Reverb and Chorus effect. Coupled with the right sequencer, that's a complete electronic music studio right there.

While MIDI offers the fullest Coral experience, OXI Instruments haven't ignored control voltage. As a compact monophonic synth voice via CV/gate, the Coral still has a lot going for it and the CV modulation options would make it at home in any modular system. You can also make use of the multitimbrality via the Part CV input, using voltage to switch between which part the CV/gate input is playing. Plug in an LFO and that makes for some very interesting melody lines.

At the time of the review I was using firmware 0.4.7, and it's not perfect. There were some odd glitches, and sometimes it got into the sort of muddle that only switching it off and on again would cure. The sample engine could be considered a work in progress. It was troubled by clicks and artefacts and felt a bit unstable. The potential is enormous, though, if it could be rendered a bit more solid. OXI wanted me to stress that they are working vigorously on the firmware and improvements, and many more features will be forthcoming.

Overall, the Coral is an amazing source of sound in a small space. It works well as a MIDI synthesizer, giving you a lot of choices, and a workable synth architecture while having some well-thought-out CV patchability. You could argue that complete synth voices, MIDI and polyphony are not really what modular is about, but the beauty of an open format like Eurorack is that we don't have to reach a consensus and can use whatever fits our particular needs. Coral can bring a lot to a small system or offer those occasional big synth sounds to a larger modular environment, and for performing, you're not having to lug a big keyboard around. *Robin Vincent*

£ £485
W www.oxiinstruments.com

Modular Profile

Matthijs Munnik

Since founding Cosmotronic along with Detlef Villerius in 2020, Dutch designer Matthijs Munnik has developed his products with all the diligent care and unerring passion one would hope for from a boutique Eurorack company. Cosmotronic's designs often take the form of elegant sound-sculpting tools, yet just as frequently deploy gloriously off-kilter circuits to catapult patches into altogether uncharted territory. The company's folio includes the side-chained distortion Peradam, described by Cosmotronic as "a compressor's twisted brother," the formidable Vortex complex oscillator and most recently the generously functional Cosmix Pro mixer (reviewed last month), which ably demonstrates how Eurorack developers must stay both creative and light on their feet in the face of today's onslaught of uncertainty; from component shortages to economic flux.

On his entry into modular

I was introduced to Eurorack some years ago by Jan Willem who runs Ginkosynthese, as we both have our workshops in the same building. He showed me the modules he had designed, and I soon ordered a couple of kits myself. That was the start of my first small system. At the time I was already building small drone pedals, circuit-bending toys and tinkering with Lunetta circuits, but Eurorack opened up a whole new space of creative possibilities.

On the story of Cosmotronic

After a while I started designing my own modules; really bread and butter stuff, but it was super exciting to be able to make a module from scratch. I never designed these with the intention of releasing them, it was really just for fun and to learn about analogue electronics, and synthesis in general. The first module I designed that I felt confident enough about releasing was Delta-V, a dual function generator with built-in VCAs. Luckily the release got a good response — except for the panel design! Admittedly the panel was a love-it-or-hate-it type deal, and I've since adopted a more minimal aesthetic. From there I've been slowly designing and releasing new modules. I feel like there's still a ton to learn and discover. Lately I've also been playing with the idea of releasing small runs of more experimental modules, since there are a lot of fun circuits I've designed that are currently still in the sketchbooks.

On his go-to modules (aside from his own!)

I really admire Mutable Instruments, not only for the great modules — which I use quite a lot in my rack — but also because of Émilie Gillet's amazing contributions to the DIY community. In a similar way I really enjoy NLC [*Non Linear Circuits*]: what an amazing collection of crazy ideas! I have built a small collection of NLC modules, and they are a lot of fun to patch. For sequencing I love the Westlicht Performer, which is just an incredible sequencer. I'm all for analogue design, but I enjoy having digital modules in the rack as well. For example the E-RM Polygogo; that's a great-sounding module in my opinion, and a really interesting interface.

On the new Cosmix Pro mixer

The first version of Cosmix was quite popular, but unfortunately became another victim of the great chip shortage. That's why I wanted to design a new mixer as its successor, and to add on some of the most requested features. This design adds performance-friendly mute buttons, a stereo aux bus and return and has the jacks moved to the top. There are also expander connectors on the back to allow for future expanders, and I hope to add even more functions later that way.

On the culture of modular

I feel the modular scene is such a diverse and welcoming community. Within the Eurorack system there is space for all kinds of ideas and experiments, and creativity is rewarded. I think one of the key reasons is the comparatively low barrier of entry, because as a designer you don't need to worry about an enclosure and a power supply. It's possible to run a small batch of a module to see if it catches on, and that gives a lot of design freedom.

I really feel like the Eurorack format is where most synthesis innovation is happening these days. And once things bloom there they start showing up in other hardware synths as well. It has been an interesting development as well to see Eurorack companies moving into hardware synths, for example the Intellijel Cascadia. *William Stokes*

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JBL PRX900 Series

Active PA Speakers

JBL's newest PRX range combines quality sound, powerful DSP and remote system control.

MIKE CROFTS

I've used JBL portable powered and passive speaker systems for a number of years, and I've always been happy with their levels of performance, reliability and value for money. So when I heard that a new PRX range had been released I readily agreed to collect a set of units and give them a try. I was offered the opportunity to visit Harman distributors Sound Technology at their impressive and wallet-tempting demo facility, where the entire PRX range is currently assembled for listening and instant comparison by selecting any combination of any units from the range.

The PRX900 series currently includes 8-, 12- and 15-inch two-way speakers (the PRX908, PRX912 and PRX915, respectively), and 15- and 18-inch subwoofers (the PRX915XLF and PRX918XLF). All models feature significantly beefier power amp packages than previous PRX speakers (2000W peak), JBL's own new drivers, new cabinet designs in polypropylene+talc (full range) or 18mm birch ply (subs), and a whole host of onboard DSP functions — all remotely controllable via Bluetooth LE using the JBL Pro Connect app, or directly from the back panel.

I've owned various JBL PRX, SRX and VRX products; the latest PRX900 range is particularly interesting as it represents a significant upgrade on the previous PRX800 range, and inherits some of the technology developed for use elsewhere within the Harman brands family. The PRX900 range is a completely new design, not just a tweaked-up version of the preceding models — in fact, as far as I'm aware, the only components that have made it through from the 800 series are the compression driver, and possibly the pole-mount socket!

Building Blocks

Every self-powered speaker is essentially an amp and a driver section, but before moving on to the clever way in which these work together, it's worth pointing out that the amp modules in the PRX900 range are much more powerful than before, and even though they are rated at 2000W they are capable of even greater output but limited to run conservatively, in the pursuit of reliability.

The drivers are of course JBL's own products, designed for the job. The HF compression driver is worthy of note, as it is the same JBL 2408H-2 1.5-inch-exit unit used in some of the high-end products such as the VRX range. I always feel more comfortable with larger-exit horns as they generally have higher power ratings and lower crossover points, and in the PRX900, they are presumably running well within their design parameters. It's also good to know that, as standard JBL models, they are available should a replacement ever be needed, although all the PRX900 boxes come with a seven-year warranty so hopefully that's not going to be an issue!

The cabinets are all-new designs, the full-range models being moulded from a polypropylene and talc mix. They feel reassuringly sturdy and I didn't detect any rattles or creaks when bumping them around, so the internal bracing and stiffening is doing its job. The subwoofer cabs are made from traditional 18mm plywood and have an attractive textured finish; if anything these are a little lighter than they look, whereas the 'top boxes' are slightly the opposite. The neat PRX family look is carried over from previous ranges, with large comfortable carrying handles and a shaped black steel grille with no externally accessible retaining screws that I could see, so there's nothing to work loose.

The Three Ps

The huge popularity of self-powered speakers reflects on their portability (everything in one box), performance

JBL PRX900 Series

From £1049

PROS

- Excellent sound.
- Easy to rig and use.
- Plenty of DSP, with remote control option via app.

CONS

- None.

SUMMARY

Powerful, scalable and easy to use, the PRX900 range has everything you could want from a portable PA system.

(amps matched perfectly to drivers) and protection (built-in limiters to avoid terminal damage). As these products have evolved, more features have been incorporated to appeal to a wider market, and the new PRX900 line is what one might call 'fully loaded' in this respect.

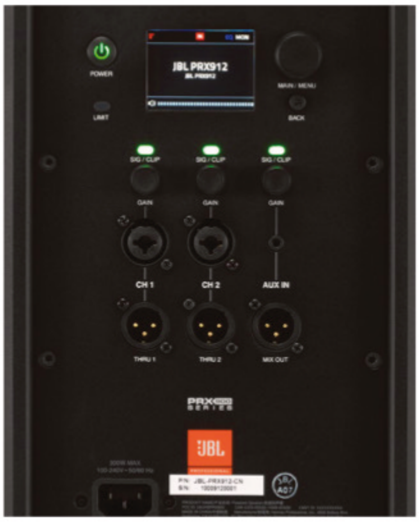
JBL products share a family home with some major pro-audio brands, and the PRX900 range benefits from the integration of features from the well-known dbx DriveRack, namely limiting with their trademark soft-knee Overeasy algorithm, automatic 24-filter feedback suppression, and direct dial-in speaker delay up to 100ms in 100-microsecond intervals. The delay feature is really useful and would be great to have in distributed speaker layouts, or simply where your mixer just doesn't have the facility. Add to all this 12-band parametric EQ on the full-range speakers and you have a pretty substantial control package.

The PRX908, 912 and 915 models have an integral three-input digital mixer with two XLR mic/line channels and a stereo channel with 3.5mm jack input. Each XLR input has a direct out, and the mixer output is also available for sending on to another system or subwoofer, with DSP settings applied. The subwoofer DSP includes cardioid presets, so setting up directional subs doesn't require any special external connections.

Appiness

The colour screen on the rear panel displays status and options, and a single data wheel allows full control of all DSP functions. The menus are easy to read and the data wheel is a straightforward turn-and-press operation, with a useful 'back' button just below it. All the functions can also be accessed using the JBL Pro Connect app, which is a free download for Android and iOS phones and tablets,. Not only does this provide a larger display of levels, settings, EQ curves and so on (much like the display screen on any full-size digital mixer), but individual speakers can be grouped and monitored within the app: up to 10 speakers can be controlled simultaneously.

You can also see more of what's going on within the app: for example, if you select the Cardioid preset on a subwoofer, the EQ screen pops up and shows what settings are being applied. It's a neat and well thought-out app that gives access to all user-controlled parameters and offers a degree of speaker control that should cover just about any deployment of the PRX speakers.



The PRX908, PRX912 and PRX915 each have a built-in mixer section, and can accommodate two mic/line signals plus an aux in on mini-jack.

In the past I have experienced some very flaky Bluetooth implementations, but all seems to be well sorted with the PRX900s, which use the latest version of Bluetooth Low Energy or BLE. Pairing is quick, easy and stable. Individual speakers are assigned an identification code, and where you have multiple devices connected, you can identify the one you are working with from the app, by making the front and rear LEDs flash on the particular speaker being addressed. Other JBL models can also be controlled using the app, and at present this includes the PRX One, Eon One MkII and Eon700.

Setting Up

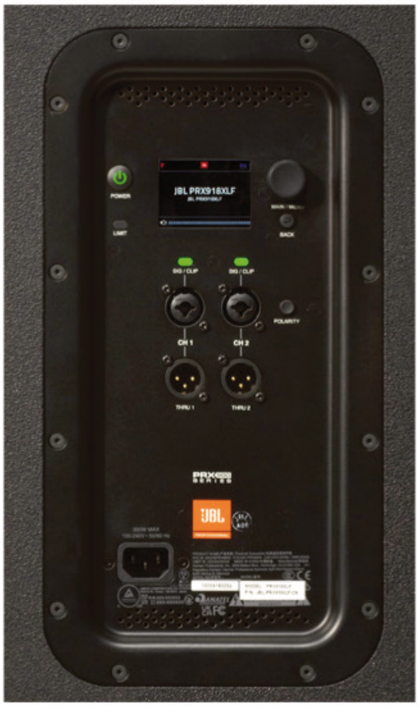
Getting the PRX900 speakers up and running was straightforward — most of the time was spent in removing them from their shipping boxes. A deliberate press (couple of seconds) on the power button lights up the display, and you're in business. There was absolutely no switch-on noise, thump or anything else, and even before connecting an input I noticed that these are very quiet indeed when at idle — with everything turned flat out on one of the PRX908s there was a very faint background noise from the LF driver, but I had to press my ear against the grille to hear it. In normal use I'd use the word 'silent'.

The units should power up in the same state as they were switched off, so I did a factory reset on all of them just to make

»

» sure I was starting from a clean sheet; this is executed from the Settings menu and required only a confirmatory press of the data wheel. After a couple of seconds of all three input LEDs flashing, the speakers were as they had left the factory, with only normal defaults set, no presets applied, and the input and master levels at minimum.

Applying a music signal to one of the XLR inputs, I raised the channel volume until I saw a hint of red LED, dialled it back a bit, then slowly raised the master level. You have to apply a fair number of turns of the rotary knobs because there's a very wide range of input control available, and it goes up and down in precise steps, all the while displayed on the screen as a dB figure. The master level setting is also reflected by a horizontal green bar across the bottom of the screen. There is no mic/line sensitivity selector on the PRX900 speakers; instead, the sensitivity auto-resolves as you adjust the input level. It's worth mentioning that the PRX900 series has some serious limiter protection built in, some of which the user can adjust and some of which is baked in to offer component protection should it ever become necessary. The non-user limiter is transparent to the operator and will only ever be activated *in extremis*, and even then it doesn't simply shut everything down and enforce a cooling-off period — my understanding is that it applies gain reduction in small, virtually unnoticeable steps on the basis of what's needed and nothing more. The limiter you can access will be somewhat familiar to DriveRack users, and the parameters you'd expect are present. In its default state (ie. after a factory reset) the make-up gain is set at zero, but the controls show that 30dB is available, and in testing these speakers to see how far they could be driven it was necessary to apply (some of) this to achieve stupidly loud output levels.



Like the mid/tops, the subwoofers feature extensive DSP, including presets for optimising them for any configuration of PRX900-series speakers.

I first set up a pair of PRX912s without subs, and ran them with a known test track. Having had the benefit of a quick demo when I picked them up I had an idea of what to expect, but in my own space I was extremely impressed with the smoothness and clarity, with a good cohesion between top, mid and bottom and no apparent gaps or dips through the frequency range. I switched styles and listened to ultra-clean, uncompressed recordings, big film scores and just about everything I could throw at them, and even at quite high levels the detail and musical balance was preserved. Hearing a crisp timpani entry, in context, in the middle of a thickly scored orchestral piece was very pleasing.

The input signal LEDs turn red to warn that the input is getting hot, but in practice I found I could drive these quite a bit harder before I noticed any hint of stress in the sound. Switching back to a nice punchy jazz/rock track, I introduced the pair of PRX915XLF subs into the equation, and this not only achieved massive low-end output but also let the PRX912s truly off the leash, with an immediate jump in overall output. I replaced the PRX912s with the PRX908s and then reduced the subs to just one; even at this reduced hardware level there was still a very punchy output, and the clean, smooth mid and top — especially noticeable on vocals — was always maintained.

Real World

After my workshop listening I was able to take the PRX rig out on a live gig to see how it performed and how easy the system was to set up and use. I had a couple of try-out opportunities, and first of all, I used the PRX912/PRX915XLF combination with my own band at an annual event. The sound requirements for this band always seem to turn out more complicated than I think they should be, mainly because it's a fairly big line-up (17 in the band including two singers) and includes a lot of brass, which can present problems in getting the vocals across cleanly, especially in the fairly lively space we play. The audience area is flat-floor, cabaret-style, and although the ceiling isn't low (as in so many hotel function rooms, for example), there are plenty of hard surfaces and not much in the way of 'soft furnishings' other than the audience members themselves.

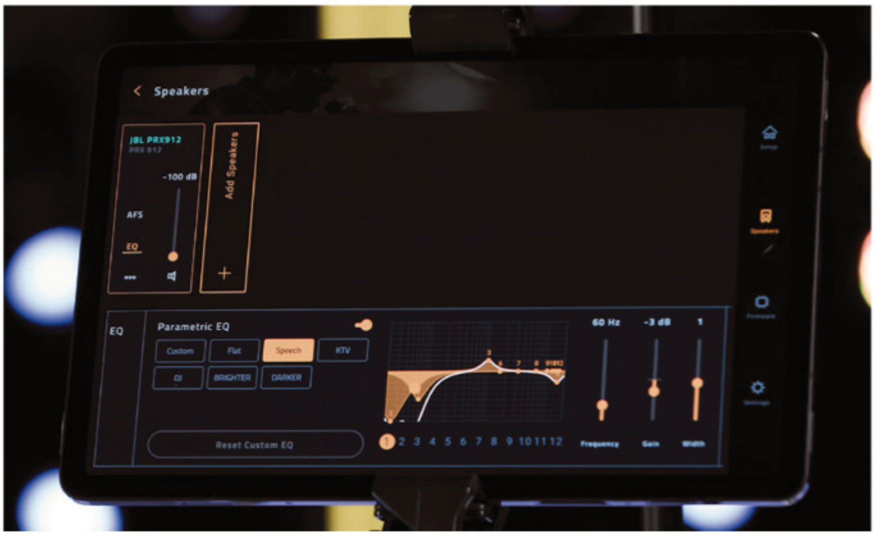
I set up one PRX915XLF each side and slightly forward of the band area, about a foot or so in from the side walls, with a PRX912 mounted on a pole above each. As the band were playing from the floor, at the same level as the audience, I made use of the tilt option on the dual pole-mount socket, angling the PRX912 boxes down so that they were pointing directly at the audience. This is a feature sadly lacking on my usual full-range speakers, so a big tick in the box straight away for the PRXs.

Although I had been playing around in the studio with the functionality offered by the JBL Pro Connect app, I was going to be driving the rig from an external mixer so I decided to go 'vanilla' as far as the speakers were concerned, and I applied a factory reset to all the PRX units, then told the PRX912s that they would be teaming up with PRX915XLF subs (a simple menu choice within the back-panel DSP) and fed the PRX912 mix outputs to them. When the speakers are set up to operate like this there is nothing else you have to do, as all the crossover and EQ settings are pre-programmed — although you can of course apply tuning and balancing as required to suit the occasion.

I ran a little bit of recorded material through the system just to check all was working as expected, then turned the subwoofer levels down quite a bit (these PRX915XLF boxes are eager beasts) until all sounded good. I spent a few minutes walking around the still-empty venue to check coverage, and the downward angle of the top speakers seemed to provide an ideal even spread throughout the roughly 20 x 20m space, with a hundred or so chairs waiting for the paying public. The band were not, mainly for reasons of logistics, going to get a proper soundcheck, so a degree of trust in the sound system was going to be necessary, and listening to recorded music in an empty room isn't a very good indication of how it will sound with 17 musicians, as a lot of direct sound will be coming direct from the stage area.

Monitors

I had intended to rig the PRX908 boxes for a comparison, but this didn't happen for two very good reasons: one, I ran out of time



All PRX900-series speakers can be remotely controlled using JBL's Pro Connect app.

and, two, I decided to use them as floor monitors. Having restored them to stock settings, I used these small but solid units laying sideways in front of the band, one for the singers and the other for general stage monitoring. The PRX full-range boxes have a very neat trick in this department, in that when you tip them over into landscape orientation, an internal sensing mechanism automatically applies some subtle DSP changes and optimises performance for this application. Although this, as with everything else, is under user control from the app and panel menus, I thought I'd just try them out with no further intervention from me, and I have to say that as far as I and the singers were concerned, nothing more was needed — they just worked very nicely and provided that much-needed clarity and cut-through that was required for vocals against the heavyweight band just a few feet behind.

Physicals

A really important part of any setting-up routine is the physical handling of the speakers, and I had, through lack of good planning, found myself all alone when it came to the part where the system had to be assembled. As mentioned earlier, these PRX boxes are solid and have a substantial feel to them, but although not in the featherweight category, they are quite easy to handle for a one-person rig.

Speaking of handles, the integral handles are extremely good and comfortable, although the shape of the PRX912 dictates that it only has one side-mounted handle, which means that to hoist it on to a pole or stand you have to support it from underneath with the non-handle hand, if you see what I mean. Two people is better, but one can manage just fine. The pole sockets on the full-range cabinets are strong and a standard 35mm pole snicks into place easily; they do not appear to need any kind of clamp screw to hold them in position once aimed. The test of good-quality, well-designed pole sockets actually comes when removing them after the gig, and the hardware on these PRX units appeared to tick all the boxes. I particularly like top handles on 'upright' speakers as this makes moving them around, especially in the back of a van, so much easier, and the PRX912 and PRX908 benefit from large, comfy fittings in this regard (note the PRX908 only has a top handle, but for a box this compact, it would be a waste of time fitting another to the side!).

PRX Performance

After all this talk about setting up, the live performance itself was remarkable only in that our drummer, who sits at the back, offered an unsolicited comment to the effect that he could "actually hear the vocals this time", by which I'd like to think that meant he could hear them better than was usually the case. I too could hear the vocals very well indeed, and not just through the PRX908s on foldback duty: my role as bandleader and conductor places me almost forward of the monitors, so I hear a lot of the main speakers too. The overall sound was very pleasing; clear, full and powerful with a real solid thump from the kick and bass, clean and un-hyped low-end power where needed (this was particularly apparent when we played a few instrumental numbers), and the vocals were the best they'd ever been, coming effortlessly over the band without ever sounding harsh.

The PRX rig had that quality of being able to fill the venue without sounding as if had just got out of bed, and from my testing/messing time in the studio I knew that it had, if called for, a whole lot more power to unleash. We received several compliments about the sound from audience members, and as these were generally from what we might call mature customers (it was a '60s-'70s-'80s night, after all) I knew that they were particularly valid for this gig.

Overall, the PRX system proved a winner on the night — a distinct step up in quality from my usual (definitely not budget) system, and I really liked the scalability aspect: although we used the PRX908s as floor monitors, they could easily have been used as front boxes in a smaller venue where we'd prefer to travel light.

Follow Spot

As a follow-on from the main live test, I did manage to use the PRX full-range speakers a couple more times in rehearsals and for providing backing amplification for a choir. My impression of the sound quality didn't change, and the PRX908 has become my new favourite as a standalone powered speaker for electric piano, for example in a pit band, or for on-stage accompaniment. With minimal adjustment from the control app, it produces a lovely even sound with a nice integration across low, mid and high piano ranges, and it's really easy to carry around and set up. I also used the PRX912 speakers in a corporate AV setting for speech-led presentation and music playback, and their smoothness and even coverage were exactly what was required.

In summary, this new PRX900 series represents a real step up from the PRX800 line. It's a bang up-to-date, well thought-out portable sound system that contains versatile and scalable components to meet just about any portable PA requirement. It looks good, it's easy to operate, has an excellent control app to access a range of useful DSP functions and more, and it not only packs a punch but sounds great doing it.

£ PRX908 £1049, PRX912 £1199, PRX915 £1299, PRX915XLF £1449, PRX918XLF £1569. Prices are per speaker, including VAT.
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Spotlight

Classic Console Preamps For The 500 Series

Several famous console manufacturers now make their preamps available 'to go'.

LUKE WOOD

Most modern audio interfaces are equipped with very capable mic preamps that can deliver a commendably clean and transparent signal into your computer, but many engineers crave the sound associated with classic recording consoles, and often still look to mixers or outboard preamps to impart some of this 'character' when tracking or mixing. There's a bewildering array of standalone preamps to choose from now, but the modular 500-series format offers a particularly convenient way to access the input sections of several classic mixing consoles. As well as taking up far less space than consoles or rackmount preamps, they're usually more affordable, too, putting them within reach of those with tighter budgets. In this article, we shine the *SOS* spotlight on 500-series options available from companies who have designed and built some of the world's most iconic studio consoles.

Neve 1073LB

Originally designed in 1970, Neve's 1073 is one of the most popular preamp circuits of all time. With the 1073LB, which we reviewed back in *SOS* December 2011 (www.soundonsound.com/reviews/neve-1073lb-1073lb-eq), the company have managed to squeeze the original Class-A circuitry, including the input and output transformers, into a single-width 500-series module. An Elma rotary switch provides up to 80dB of gain in 5dB steps, and -20 to +5 dB of fine adjustment is offered by a variable Trim control — so the module offers up to 85dB of gain in total, and is capable of delivering the harmonic saturation for which the 1073 is renowned. Switchable impedance settings of 300Ω and

Neve's 1073LB preamp.

The Neve 88R LB.

The 312 — API's authentic vintage preamp.

1.2kΩ provide useful tone-shaping options with vintage and modern microphones, and some other handy features include a polarity switch and selectable front-panel or host-rack input connections.

£ £882 including VAT
W www.ams-neve.com

Neve 88R LB

The 88R LB (reviewed in *SOS* November 2012: www.soundonsound.com/reviews/neve-88r-lb) features the preamp circuitry employed in another Neve classic, the 88RS large-format console. A transformer, manufactured to the exact specification of the Marinair designs used in early Neve mixers, is used to balance the input stage, though as there's no output transformer this time, the character tends to be more subtle than that of the 1073LB. Up to 70dB of gain is available, and the module is capable of accepting microphone, line-level and high-impedance instrument signals, whilst a Regeneration function expands on the capability of the original console preamp by allowing line-level and instrument sources to be routed through the input transformer — an appealing feature for musicians and engineers alike. A variable high-pass filter ranges from 31.5 to 315 Hz, and there's polarity inversion, front/rear input selection, a 20dB pad and a ground-lift switch.

£ £702 including VAT
W www.ams-neve.com

API 312

Based on API co-founder Saul Walker's early designs, the 312 (reviewed in *SOS* November 2020: www.soundonsound.com/reviews/api-312) remains one of the world's most sought-after microphone preamps, having been a key element in API's consoles throughout the 1970s. With a circuit design featuring the company's 2520 discrete op-amps and proprietary input and output transformers, the module aims to offer access to the classic API sound for rather less than the cost of a mixing console. Up to 69dB of gain is available, and the module is equipped with a 20dB pad and polarity inversion buttons, as well as an option to tap the output transformer at a 1:1 rather than 1:3 ratio, reducing the level by about 10dB and making it easier to accommodate hot signals. The front panel also sports a vintage-style analogue VU meter.

£ £745 including VAT
W www.apiaudio.com

API 512v

Another API offering, the 512v (see *SOS* December 2016: www.soundonsound.com/reviews/api-512v-mic-preamp) has much in common with the 312: its discrete circuitry also employs the company's proprietary transformers and 2520 op-amps, so again it promises to deliver the rich-sounding, musical tone associated with API's early consoles. The 512v, though, provides some useful additional features, including a variable output level control that allows you to drive the preamp harder in search of character, without overloading the next device in the chain. Up to 65dB of gain is available, and it is equipped with

»

API's 512v delivers the classic API sound while adding some useful features.

SSL
VHD+
SIGNAL
MIC PRE
DRIVE
VHD
GAIN
TRIM
HPF
48V
PAD
HI-Z
Ø
LINE IN / DI
VHD MIC PRE

SSL's VHD+ offers controllable character.

S1P TUTTI
50th YEAR EDITION
SIG
GAIN
POL
TRIM
48V
INST
FADER
MIC/INST
SUNSET SOUND

Sunset Sound's S1P Tutti: a preamp that's as colourful as it looks!

» a 20dB pad, a polarity inversion button and the same output transformer ratio switch as is found on the 312. There's also a front-panel XLR/TRS combi socket that can accept instrument or line-level signals.

£ £995 including VAT
W www.apiaudio.com

SSL VHD+

The VHD+ features SSL's SuperAnalogue preamp circuitry, which offers the clean, transparent sound of the preamps in their modern consoles, but also incorporates the Variable Harmonic Drive (VHD) circuit that was originally introduced in their flagship Duality console. (We've not reviewed the 500-series version separately, but you can find out more about the preamp design in our review of SSL's XL-Desk console in *SOS* January 2015: www.soundonsound.com/reviews/ssl-xl-desk). In essence, the idea is that the user can turn the Drive control to add as little or as much of the 'grit' associated with SSL's older consoles as is desired, ranging from a tube-like warmth to obvious distortion. The module provides up to 75dB of gain, and a ±20dB Trim knob is present to keep the output level under control. A variable 18dB/octave high-pass filter (15 to 500 Hz) is joined by a 20dB pad and a polarity inversion button. Instrument and line-level sources can be connected to the front of the module, and the Hi-Z switch raises the input impedance to a guitar pickup-friendly 1MΩ.

£ £598.80 including VAT
W www.solidstatelogic.com

Sunset Sound S1P Tutti

The S1P Tutti (reviewed in *SOS* November 2014: www.soundonsound.com/reviews/sunset-sound-s1p-tutti) is based on the input section of the custom API/DeMedio consoles in Hollywood's legendary Sunset Sound studios. The current module is the result of a collaboration between the studio's owner Paul Camarata and Paul Wolff, the designer who created API's original Lunchbox rack. True to the original design, the module employs dual John Hardy 990c op-amps and a pair of custom Cinemag nickel input and output transformers, and it's said to be sonically indistinguishable from the original units. Up to 60dB of gain is available in 10dB steps — this can be fine-tuned with an additional ±6dB trim control — and an output control allows the overall level to be pulled back when driving the preamp harder. A polarity switch is provided, and the module will accept a microphone input either via the host rack's input or through the front-panel combi socket, which can also accommodate high-impedance instrument signals.

£ $1195
W https://sunsetsoundstore.com

TRIDENT
PREAMP
OUT (VU)
+48V
MIC
LINE
LPF
Polarity
HPF
LINE (HI Z)
80B-PRE
TRIDENT

The new Trident 80B-500 includes variable high- and low-pass filters.

Trident 80B-500

The popular Series 80B consoles, designed by Trident Audio Developments in the late 1970s, were popular for many reasons including their clean-sounding, low-distortion preamps, which offered bags of headroom. The recently announced 80B-500 is said to be a faithful recreation of that preamp circuit for the 500-series format. It has a Lundahl transformer-balanced mic input and electronically balanced line input (with separate gain controls for each), and a Lundahl transformer-balanced output. A front-panel high-impedance input hooks your instrument signals up to the line amp circuit. The 80B-500 can accommodate signal levels from -60dBu to +15dBu with no need for a separate pad, and while it uses pots rather than switches (for up to 65dB of gain), Trident say there's none of the 'gain cramping' exhibited by cheaper pot-based designs. Adding considerably to this model's versatility is the pair of variable high-pass (30Hz to 350Hz) and low-pass (2KHz to 20KHz) filters. There's also a polarity invert button, switchable 48V phantom power and an eight-LED meter. Trident have also recently announced a 500-series version of their A-Range preamp, though full details weren't yet available when going to press.

£ $649
W tridentaudiodevelopments.com

Chandler TG2-500

No discussion of classic consoles would be complete without a mention of Abbey Road Studios. With the introduction of eight-track recording in the late 1960s pushing the studio's REDD consoles to their limits, Abbey Road's engineers joined forces with the Central Research Laboratories at »

Neve 1073OPX

ADAT | ANALOGUE | DANTE | USB

58.
20.
MIC LN DI
IP
FNT
80
7
8
SIG
96 176 192
WCK USB DANTE ADAT LOCK COMMS
USB
DANTE ADAT
Neve
1073OPX
MONO
1/2 5/6 MON
3/4 7/8 DMON
HP LEVEL

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DESIGNED AND CRAFTED IN THE UK BY NEVE® ENGINEERS

CHANDLER LIMITED
COARSE GAIN
FINE GAIN
OUTPUT
TG 2-500

The Chandler TG2-500 packs an Abbey Road/EMI preamp into the 500-series format.

» EMI Hayes to develop a replacement. Their efforts resulted in the EMI TG12345. Officially licensed by EMI and Abbey Road Studios, Chandler's TG2-500 (reviewed in *SOS* January 2015: www.soundonsound.com/reviews/chandler-limited-tg2-500) recreates the TG12428 preamp section of that rare and coveted console. The unit employs discrete, transformer-balanced amplifiers, and when driven hard exhibits some distortion and a rise in its high-frequency response, helping to deliver the warm and punchy sound associated with the console. A Coarse Gain switch offers 50dB of gain in 5dB steps, whilst a Fine Gain pot provides ±10dB of variable gain. A polarity inversion switch is present, and users are able to choose between input impedances of 300Ω and 1.2kΩ.

£ £1049 including VAT
W https://chandlerlimited.com

Honourable Mentions

Rupert Neve Designs
511
MIC PRE + SILK
POWER
TRIM
GAIN
48V
LEVEL
HPF
TEXTURE
SILK

The Rupert Neve Designs Portico 511.

The late, great Rupert Neve was involved in several companies over the course of his life and the last of them was **Rupert Neve Designs** (RND), which he founded with Evelyn Neve and Joshua Thomas in 2005. They created the 5088, a magnificent modular console — a real modern classic. The 5088's preamps are too big to condense into the 500-series format, not least because of the enormous transformers that are a key part of its sound. You can find them in the company's larger Shelford modules, but RND do offer a different Rupert Neve-designed preamp for the 500 series. Called the Portico 511 (reviewed in *SOS* May 2018: www.soundonsound.com/reviews/rupert-neve-designs-500-series), it couples a clean, transparent preamp with their Silk feature. A development of Rupert's Amek designs, Silk increases the amount of harmonic distortion present in the mid and high frequencies, to emulate the sound of Neve's early Class-A designs. A Texture control allows users to determine how much distortion is introduced, with the results ranging from almost inaudible to well beyond the levels found in vintage units. Up to 66dB of gain is provided in 6dB steps, with some fine-tuning possible thanks to a ±6dB Trim control. A continuously variable 12dB/octave high-pass filter (20-250 Hz) is present, along with a polarity inversion button.

Electrodyne currently manufacture the **501** preamp, which is based on the original engineering notes for their vintage consoles, which were common in high-end studios during the late '60s and early '70s. **Aurora Audio** offer the transformer-coupled Class-A **GT500** module, as a compact alternative to their **Sidecar** console. And **Alice** also offer three Ted Fletcher-designed modules, the **503**, **504** and **501T**, each of which is based on the circuitry used in the company's studio and broadcast consoles.

Some other companies who offer recreations of units that are now out of production include **H2 Audio**, who produce the **Helios 2128**, which recreates the preamp circuitry used in the legendary Helios 0011 console; and **Spectra**, whose **1964 STX100** module provides a modern alternative to the **Model 110** preamplifier section of the Spectrasonics console at Ardent Studios in Memphis. There's also the **DAV Electronics BG501**, designed and built by Mick Hinton, an ex-Decca Studios engineer whose designs were used in much of that legendary studio's equipment. There remain plenty of classic consoles with no obvious 500-series equivalents, though — names such as Studer, Cadac, Calrec, MCI, Neotek and Amek spring to mind — so there seems to be plenty of scope for more modules in this market!

501
GAIN
OUT
Z
-20
ø
48
DI
INST

Though less well known than Neve, SSL or API, Electrodyne consoles were highly desirable.

Additionally, there's a wide variety of modules available that are more loosely inspired by the circuitry and sounds of classic consoles, either offering alternative takes on popular models or providing a modern alternative to designs that are no longer in production. Notable examples of the former include the likes of **Heritage Audio**'s **73 JRII**, the **Golden Age Project Pre-573** and **BAE**'s **1023L**, all with roots in vintage Neve designs, whilst products such as the **Warm Audio TB12 500** and **BAE**'s **312A** and **312B** modules draw their inspiration from early API designs. For those who have soldering skills, companies including **CAPI** (Classic Audio Products, Inc) and **Sound Skulptor** offer high-quality DIY kits based on API and Neve preamps, among others. ■■■

UF1
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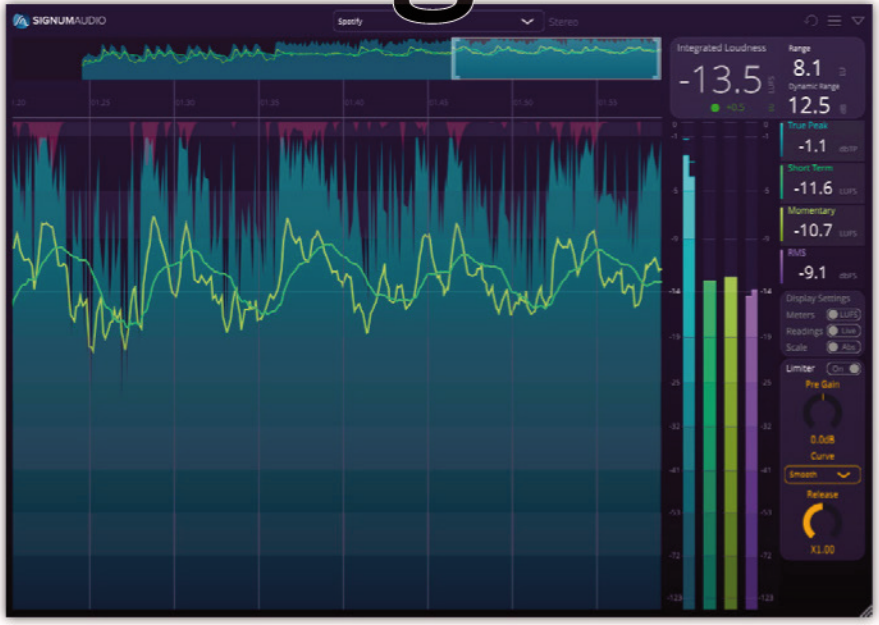
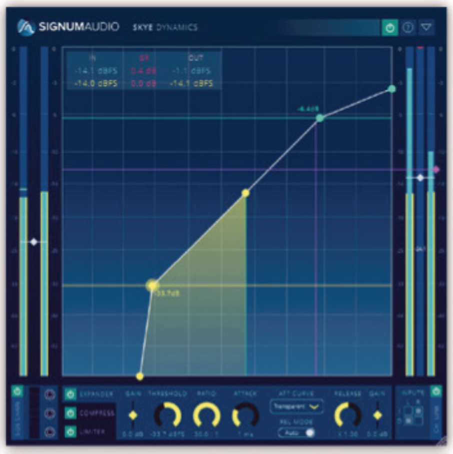
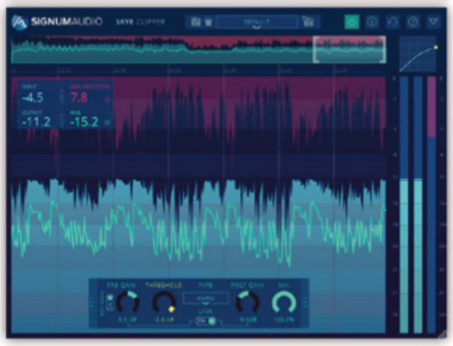


Integrated Loudness -13.5 LUFS
Range 8.1
Dynamic Range 12.5
True Peak -1.1 dBTP
Short Term -11.6 LUFS
Momentary -10.7 LUFS
RMS -9.1 dBFS
Display Settings
Meters LUFS
Readings Live
Scale Abs
Limiter On
Pre Gain 0.0dB
Curve Smooth
Release X1.00





Worth £1251

UK-based Signum Audio are specialists in DSP engineering and plug-in development, and their products all have loudness and dynamic range in mind. The company's mission is to create cutting-edge pro-audio software, designed and engineered to the highest standards and incorporating pioneering DSP algorithms while employing clean, workflow-driven, feature-rich user interfaces. Signum Audio's plug-ins are compatible with VST/VST3, AU and AAX hosts, running on Windows or macOS, and work in formats up to 7.1.4 surround.

For this month's *SOS* competition, we've teamed up with Signum Audio to offer you the chance to win one of three premium Signum Audio plug-in bundles worth £417 each, comprising the complete BUTE Loudness Suite 2 Surround, plus the comprehensive SKYE Dynamics Surround processor and the company's new SKYE Clipper Surround plug-in.

The BUTE Loudness Suite 2 includes three plug-ins: True Peak Limiter, Loudness Analyser and Loudness Normaliser. These processors are clean yet comprehensive, with a vast array of presets making it easy to deliver loudness-compliant audio for almost any broadcasting or streaming platform. Other features include a completely interactive GUI with smart history overview, a custom alert system, smart auto-metering, DAW sync and a compact readout view, to name a few.

SKYE Dynamics, meanwhile, is a three-in-one multi-stage dynamics processor comprising a compressor, expander and limiter, all of which are easily configurable and accessible from the slick single-page user interface. Bring your sound to life with features such as an advanced adaptive auto-release, variable saturation, various channel-linking options, and a selection of five envelope presets for each processor.

Finally, SKYE Clipper features Signum Audio's proprietary real-time anti-aliasing algorithm, which the company claim "drastically reduces the amount of aliasing artefacts produced by clipping". SKYE Clipper lets you choose between different clipping styles to help control peaks, add loudness to your mix, or to add saturation to your audio. This means it can be used in a variety of scenarios like taming a snare, bus processing, or to add texture to a synthesizer. It includes seven different clipping types, with a sleek, feature-rich and yet simple user interface for a streamlined workflow.

To be in with a chance of winning one of these fantastic software bundles, simply follow the URL shown and answer all the questions there, by Friday 28th July 2023. Good luck!

To enter, please visit:
https://sosm.ag/signum-comp-jul23

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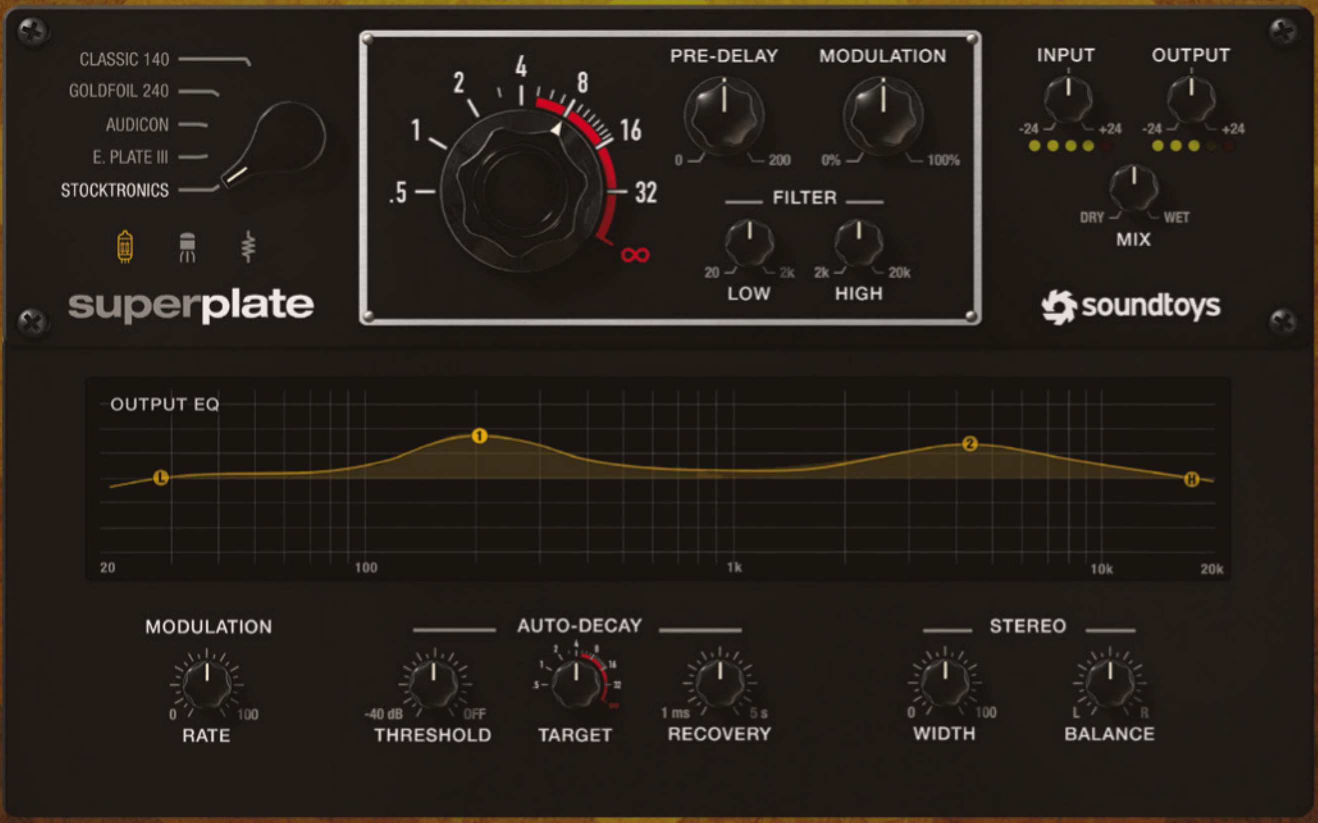
W www.signumaudio.com

NEW!

Introducing SuperPlate

AN EXPANSIVE REVERB EXPERIENCE


CLASSIC 140
GOLDFOIL 240
AUDICON
E. PLATE III
STOCKTRONICS
superplate
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PRE-DELAY
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-40 dB
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THRESHOLD
TARGET
1 ms
5 s
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STEREO
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100
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R
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SuperPlate brings the unique tonal character of five classic electromechanical plate reverbs into your studio – without the massive investment and back-breaking weight of the originals. Its advanced algorithms capture a level of unmatched realism and versatility and create luscious, immersive effects.

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soundtoys

$149 MSRP

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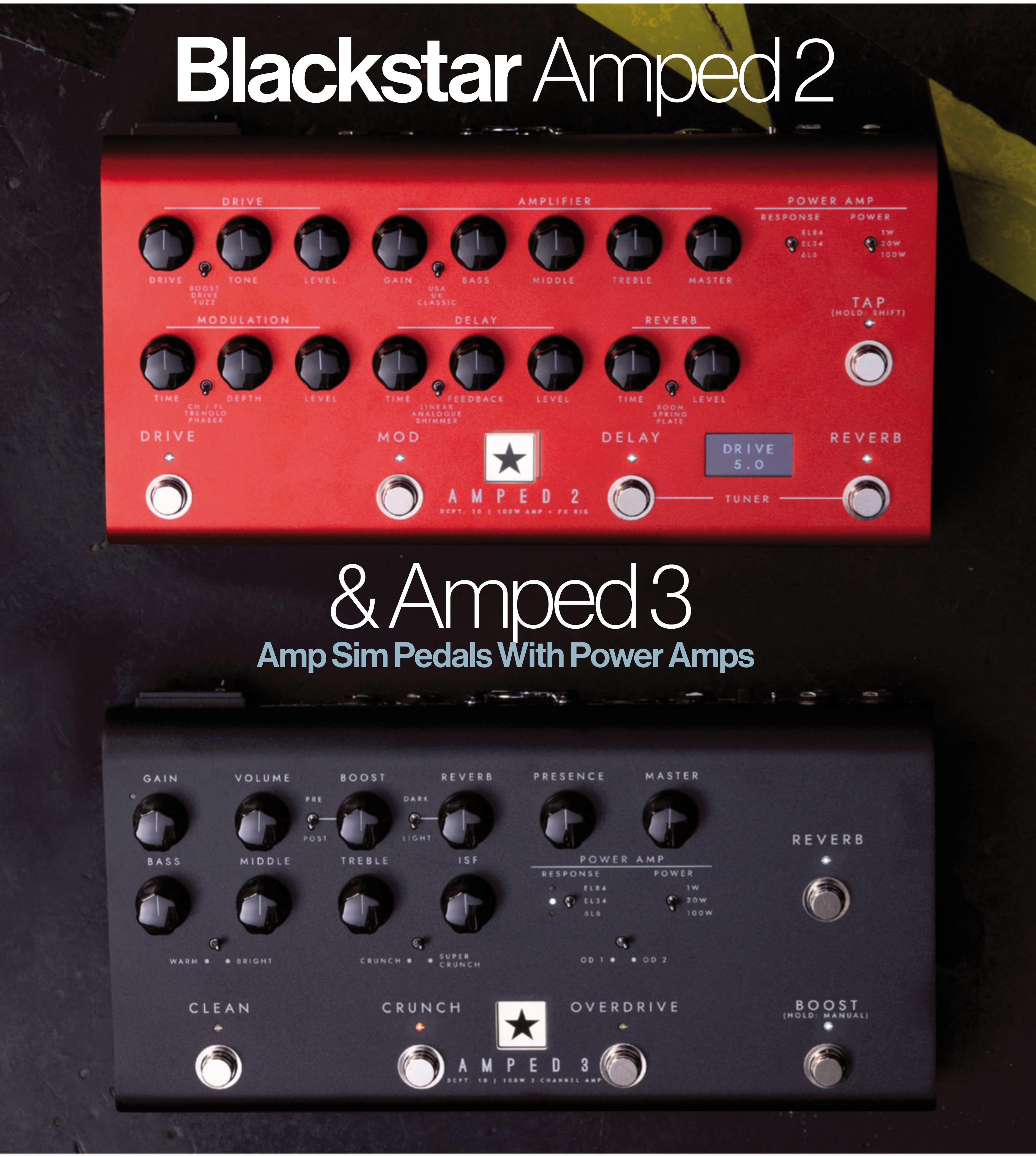
Free 30-day trial at soundtoys.com

Blackstar Amped 2 & Amped 3

Amp Sim Pedals With Power Amps


DRIVE
AMPLIFIER
POWER AMP
RESPONSE
POWER
EL84
EL34
6L6
1W
20W
100W
DRIVE
TONE
LEVEL
BOOST
DRIVE
FUZZ
GAIN
USA
UK
CLASSIC
BASS
MIDDLE
TREBLE
MASTER
TAP
(HOLD: SHIFT)
MODULATION
DELAY
REVERB
TIME
DEPTH
LEVEL
CH / FL
TREMOLO
PHASER
TIME
FEEDBACK
LEVEL
LINEAR
ANALOGUE
SHIMMER
TIME
LEVEL
ROOM
SPRING
PLATE
DRIVE
MOD
DELAY
DRIVE
5.0
REVERB
AMPED 2
DEPT. 10 | 100W AMP + FX RIG
TUNER
GAIN
VOLUME
BOOST
REVERB
PRESENCE
MASTER
PRE
POST
DARK
LIGHT
BASS
MIDDLE
TREBLE
ISF
POWER AMP
RESPONSE
POWER
EL84
EL34
6L6
1W
20W
100W
REVERB
WARM
BRIGHT
CRUNCH
SUPER CRUNCH
OD 1
OD 2
CLEAN
CRUNCH
OVERDRIVE
BOOST
(HOLD: MANUAL)
AMPED 3
DEPT. 10 | 100W 3 CHANNEL AMP

These two stompbox amps may look superficially the same and do have quite a lot in common, but each one seems to have been optimised for a different type of user.

DAVE LOCKWOOD

Just as I was finishing my review of Blackstar's Amped 2, the Amped 3 arrived. Both are products of their Dept. 10 division, and are digital guitar-amp simulation floor units, with integrated 100W Class-D power amps. The primary differentiation between them is that Amped 2 (the red one) includes a number of effects, but has no preset save/recall facility, whilst Amped 3 has no effects apart from reverb, but will allow you to store a favourite setup of clean, crunch and lead channels on board. Both, however, can access Blackstar's free Mac OS/Windows Architect software, and this allows you to select and fine-tune a whole range of CabRig speaker and mic simulations that can be output instead of, or as well as, the real-speaker signal from the power amp. You can also take the speaker-sim output directly from the onboard USB-C connection.

Both units therefore span a whole range of uses, from feeding a real speaker on stage, either miked up or sending a speaker-emulated DI signal to front-of house, through to silent recording in the studio using just the USB-C digital output straight into your DAW without even needing an audio interface.

Blackstar Amped 2 & 3

£539/£499

PROS

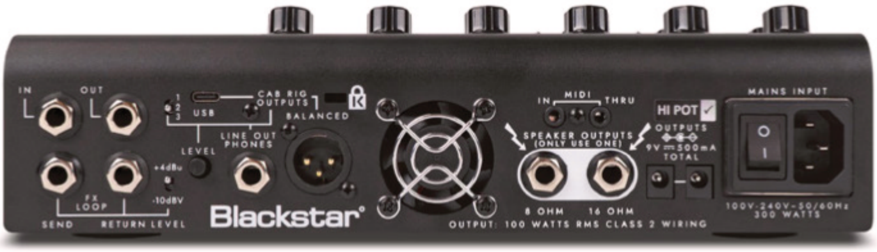
- High-quality amp modelling.
- Very ruggedly built.
- Built-in power amp works well.
- CabRig offers great DI recording tones or FOH feed.
- Can be used as an audio interface.
- An amp-like response to drive pedals.

CONS

- No smartphone version of Architect software.
- Amped 3 lacks a tuner.
- Amped 2's onboard drive pedals a little limited.

SUMMARY

A choice of two great all-rounders, whether you want a single amp with effects, or a channel switcher. Ideal for a compact lightweight stage amp, or a studio DI solution... or both.



The two devices have identical rear panels. Though necessary, the cooling fan is quiet enough that it's not problematic even for desktop recording.

Whilst the tone creation is all DSP, the Class-D power amp is a current-feedback design with a low speaker-damping factor, rather like a real valve/tube output stage, giving it some of the resonance and character of the latter when connected to a real guitar cab. It certainly works: it feels very responsive whilst delivering its full 100W into either an 8Ω or 16Ω load, and with a choice of power scaling down to 20W or even just 1W. If I've learned anything from these devices, it's that Class-D power amps don't have to be the noisy, hissy beasts that they so often are, and that just 1W into an efficient speaker can be very loud; I've done gigs where using any amount of gain with the 1W setting would have had FOH asking me to turn down! The amps are fan-cooled, and the fan runs all the time, but it is decently quiet and not really too intrusive, even when used on a desktop for direct recording.

In addition to the power scaling, both units offer a choice of simulated output tube responses: the 6L6 used by Fender and many USA makers, the EL34 most readily associated with Marshall, and the EL84 historically associated with Vox in the UK, but it's also favoured now by many boutique makers in the USA. These certainly provide audible differences, with the 6L6 sounding wide-band and balanced, the EL34 having plenty of rich midrange, and the EL84 offering chime when clean and getting some chewy-edged compression when driven harder.

Amped 2

Although it is the model with the effects on board, the Amped 2 is conceptually the simpler of the two models: whatever you dial up with the top-panel controls and switches is what you are going to hear. The Amplifier section has a switchable choice of basic amp type: 'USA' or clean Fender-ish; 'UK', which is Vox-ish; and 'Classic', which sounds rather like a hot-rodded Marshall. A standard Gain and Master setup allows you to balance out sensitivity and output volume. On its own, I found the USA-clean perhaps a little too clean, even at maximum gain, and found some warmer clean tones in the heavier drive channels with the gain backed off and the guitar volume turned down a little. A bit of drive and boost from a classic, old Fulldrive pedal into the clean channel, however, seemed to do exactly what I've always found it to do with clean Fender amps, so maybe that is just accurate modelling! Both Amped models seem to take pedals very well, even overdrive pedals that rely mainly on raising the input level. The Bass, Middle and Treble tone stage does what you'd expect, with plenty of range and a familiar degree of interactivity. The operation of the tone controls also seems to have been tweaked to work most effectively with each of the different amp voicings. I am not sure if that extends to circuit-correct positioning of the tone stack earlier for the Fender-type voicing and later for the more Marshall-like voicing, but the EQ always seems to be able to do what you want.

The effects divide into Drive, Modulation, Delay and Reverb sections, with each having a dedicated footswitch. Drive gives you a pre-selectable choice of Boost, Drive and Fuzz, with Drive, Tone and Level controls replicating what you'd find on most pedals. Similarly, the Modulation section offers a choice of Chorus/Flanger, Tremolo and Phaser, governed by Time, Depth and Level controls. No surprises in the Delay section: a choice of Linear, Analogue and Shimmer. Well, some surprise: will 'Shimmer' still be an effect everyone wants this time next year? I guess we'll find out. Delay controls are Time, Feedback and Level, which brings us to the Reverb section, with just Time and Level controlling a choice of Room, Spring and Plate simulations. The Delay section also has a Tap Tempo footswitch, which doubles as a Shift function: press and hold and you can adjust the 'hidden' Presence control using the Treble knob, aided by a small numerical screen readout. Make a mental note of your Treble setting before you do so, however, as it'll probably end up in a different place

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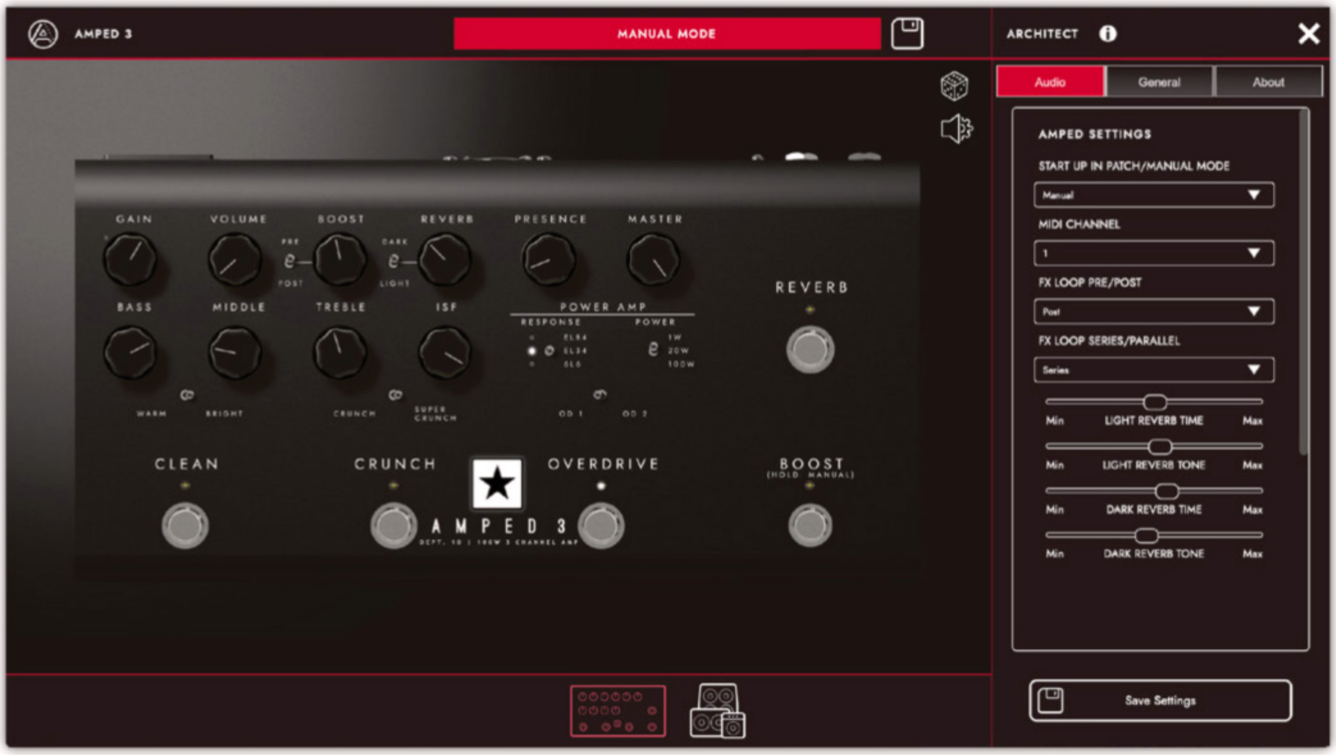

AMPED SETTINGS

START UP IN PATCH/MANUAL MODE: Manual

MIDI CHANNEL: 1

FX LOOP PRE/POST: Post

FX LOOP SERIES/PARALLEL: Series

Min LIGHT REVERB TIME Max

Min LIGHT REVERB TONE Max

Min DARK REVERB TIME Max

Min DARK REVERB TONE Max

The Architect software offers much more detailed control over your tone than you get from the hardware controls.

after you've used it to tweak the Presence. The little screen is also used as a display for the onboard chromatic tuner, activated by pressing the Reverb and Delay footswitches at the same time. Now, I have size 11 feet, but there is no way I can press both of these at once without contriving to come at them from the side, which is not a particularly elegant on-stage look for something that you might be doing rather a lot — in the kind of gigs I do, at least, I'll activate a tuner more often than any other single effect. Surely this would be better done with a simple press-and-hold on, say, the Reverb footswitch?

Of course, there's a good reason why Blackstar haven't done that: the Reverb, Delay and Modulation footswitches all already have secondary press-and-hold functions, activating a sort of Freeze or Emphasis effect that creates a greater effect intensity until you release the switch. It certainly works to give you a bit of extra expressive control, but I can't say that I would use it often enough in a performance to not prefer the greater functionality of easier access to the tuner. The Drive footswitch doesn't have a secondary function: how about putting it there with a firmware update?

Architect Software

The Architect software gives you a number of tweaks that aren't on the front panel: reverb tone can be adjusted to taste and saved to the unit, and delay can be set up to be before or after the amp stage. You can also use it to set up and store three different CabRig virtual cab and mic combinations within the unit itself, selectable from a hardware back-panel switch. This can be used to give you options in a live sound setting: select a 4x12 if you are going to be using mainly the Classic amp, or 1x12 combo-type speaker if you are going for a clean Fender type vibe, or perhaps three slight variations of the same cab and mic setup, to allow you to adapt to the environment. In a studio setting, where it's easy to access the Architect software permanently, you can dial up any cab you want from the wide range on offer at any time. Very nice they are, too, enhanced by a choice of several different types of room ambience, if you want, which helps make everything feel a bit more real. The CabRig speaker emulations are described as being "next-generation DSP-based", rather than just impulse responses.

The Amped 2 and 3 models both feature an effects loop, hardware switchable for +4dBu or -10dBV operation, allowing it to match pedal effects or 'pro-level' rack gear. Using the Architect software, the loop can also be set to series mode, with the dry signal also passing through the loop, or parallel, with the external processor sending an 'effect-only' signal back into the return, for mixing with the internal dry path. To further aid pedal usage with these, you also have a pair of 9V DC outputs with a total of 500mA current capacity shared across both outputs.

There's also a MIDI input and thru connection on both models, courtesy of a pair of supplied mini-TRS to female cable-mount MIDI sockets. For a while, this

»

Multi-channel USB audio

The USB-C connection offers four independent, simultaneous outputs and a stereo return, making it very easy to keep all your post-production options in play by recording separate processed and clean (straight from the guitar) versions of your performance on their own channels. Output channels 1 and 2 carry the CabRig left and right channels with power-amp, speaker cab and mic emulation, as well as any room simulation selected, with the signal taken post-Master volume control. Channel 3 has the sound of the preamp voices and EQ, taken before the reverb and without any speaker emulation so it can be paired with power-amp/cabinet emulation plug-ins within your recording software/DAW. Channel 4 features the direct signal from the guitar, unaffected by any of the controls or processing, ready for reamping use. The two channels coming back from the computer carry the software's left and right outputs for monitoring via the XLR (mono) or line out (stereo) CabRig outputs.

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LONDON

SOUND ON SOUND

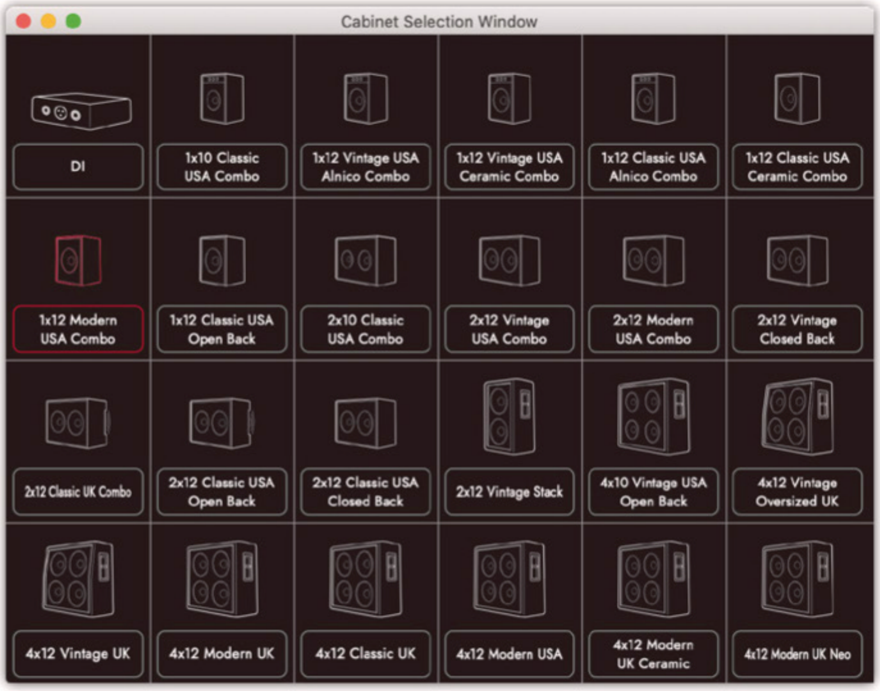
» had me looking for a way to save whole setups, for recall via an external MIDI Program Change message, even if you can't do that from the unit itself, but that's not how Amped 2 works. Instead, you can access more or less anything via MIDI Continuous Controller data. If you've got something in your rig that will allow you to send user-set CC numbers from a pedal, for example, you could have real-time control of your delay mix, or input gain. You can even access the channel voice setting and the 'virtual tube type' via MIDI. This level of deep controllability does seem slightly at odds with a product that otherwise seems like the perfect 'grab and go' all-in-one amp and effects unit. Especially one without preset recall and therefore probably aimed at the kind of player who is going to dial in their favourite virtual amp setting for the gig and work with it just as they would with a real tube amp. I know that's how I'd use this rig, because I am that kind of player. Still, I'd hesitate to criticise any maker for offering too much facility, even if I can't see it being used much!

Amped 3

If you set Amped 3 to Manual mode you've got, more or less, an Amped 2 but with none of the effects except reverb. You've still got a footswitchable Boost facility, selectable to go before the amp stage, giving up to 12dB of extra drive, or after, allowing you to boost the actual output level by 6dB; and you've also got a physical (no shifted-function access) Presence control. Bass, Middle and Treble controls here are augmented by Blackstar's Infinite Shape Feature (ISF). This re-voices the tone stack to be a bit more British, with a warmer, looser midrange emphasis; or a bit more American, so tighter, faster and with a bit more top and bottom, in very general terms. ISF actually works really well at allowing you to quickly get the basic character you are seeking, before fine-tuning using the tone controls, or *vice versa*. A single Reverb control can be switched to Dark or Light from the hardware, but there are more parameters available in Architect.

ALTERNATIVES

There aren't that many similar amp sims with built-in power amps that I can immediately call to mind: the **BluGuitar Amp 1** variants have an analogue preamp and a nano-tube power stage, the **Victory** pedals have a tube preamp, and the **H&K AmpMan** is all analogue, so maybe Blackstar have this particular niche all to themselves.


Cabinet Selection Window
DI
1x10 Classic USA Combo
1x12 Vintage USA Alnico Combo
1x12 Vintage USA Ceramic Combo
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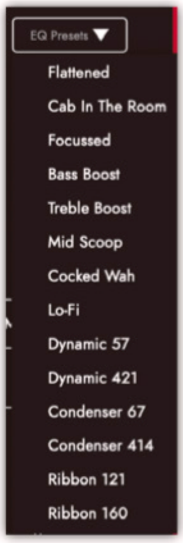
1x12 Classic USA Ceramic Combo
1x12 Modern USA Combo
1x12 Classic USA Open Back
2x10 Classic USA Combo
2x12 Vintage USA Combo
2x12 Modern USA Combo
2x12 Vintage Closed Back
2x12 Classic UK Combo
2x12 Classic USA Open Back
2x12 Classic USA Closed Back
2x12 Vintage Stack
4x10 Vintage USA Open Back
4x12 Vintage Oversized UK
4x12 Vintage UK
4x12 Modern UK
4x12 Classic UK
4x12 Modern USA
4x12 Modern UK Ceramic
4x12 Modern UK Neo

The 'next-generation' DSP-powered virtual speaker cabinets are a real selling point.

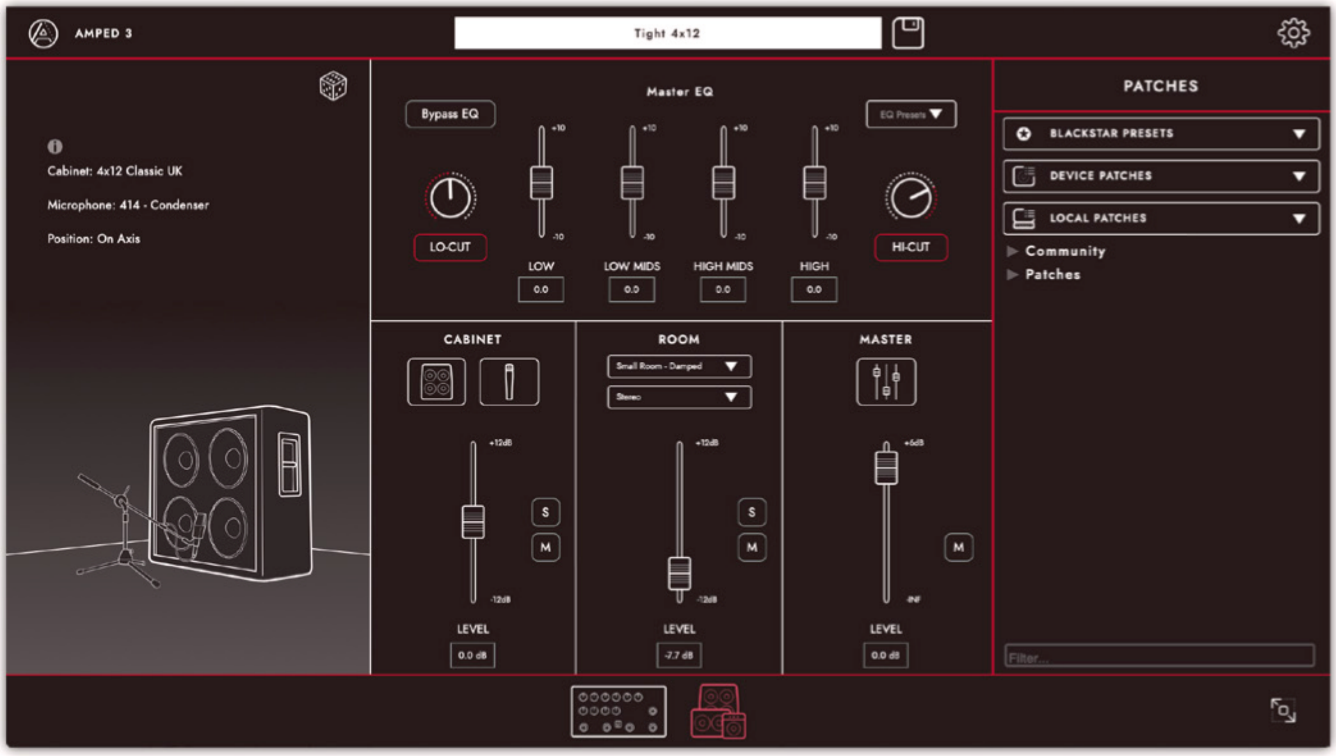
It's the same power amp in both models, so you've got the same 100W, 20W and 1W power scaling, and the same virtual 6L6, EL34 and EL84 voicings, but this one has a whole other mode of operation in the form of Patch mode. The three footswitches labelled Clean, Crunch and Overdrive probably tell you exactly what's going on here, and operation in Patch mode couldn't be simpler. Just choose a channel, dial up the settings for the sound you want and press and hold the footswitch until the LED blinks. Select the next channel and repeat. Commendably obvious and with no need to involve the software app, although of course you can if you want to access something beyond the hardware controls. Step on a different channel switch for instant recall of its stored sound. The controls won't now represent the actual settings, but you can still find their stored values: rotate any of them and look for the little Recall Indicator LED in the top corner to briefly flash. That's where the knob was set. Of course, if you are in the studio and connected to the app, you'll see an accurate graphic depiction of the knob positions as used by the saved preset, and they'll update on screen as you move them or change presets.

Clean has a choice of Warm or Bright; Crunch has... Crunch and Super Crunch; and Overdrive has OD1 and OD2 variants. Like the Amped 2 model, the Clean channel stays fairly clean at all settings, although you can push it into something a bit richer-sounding with lots of pre-boost, while Super Crunch gives you a lot of scope in the mid-gain channel: certainly enough to be an alternate lead channel. But the Overdrive channel, especially in OD-1 mode, now that's a real star! Blackstar say this is one of the highest-gain circuits they've ever developed and I daresay it is, but it is also much more than that. This is a channel that cleans up surprisingly well when you back off the guitar's volume control, giving you a whole range of nuanced tones between warm semi-clean, and searing, effortless lead tones, with everything in between, just from your pick/fingers and volume setting. Very articulate and one of the sweetest high-gain sounds I've heard coming from something that some people might hesitate to call 'a real amp'.

Of course, it is a real amp, as much as anything else these days, and I was using a real speaker, but the real surprise was finding I could match the sound, and perhaps even exceed it, using the


EQ Presets
Flattened
Cab In The Room
Focussed
Bass Boost
Treble Boost
Mid Scoop
Cocked Wah
Lo-Fi
Dynamic 57
Dynamic 421
Condenser 67
Condenser 414
Ribbon 121
Ribbon 160

The EQ presets for different virtual mic types seem to have been very nicely judged.



Tight 4x12

Cabinet: 4x12 Classic UK

Microphone: 414 - Condenser

Position: On Axis

Another view of the Mac/Windows Architect control utility, revealing more of the parameters that can be tweaked and saved to the pedal as presets.

virtual cab and mic combinations in the Architect software. These are seriously good, with a considerable amount of useful variety of choice available, and go a long way towards making these pedals the versatile, multi-use all-rounders they are — studio centrepiece, 'fly-rig' lightweight amp, compact backup rig, and more. The XLR output is mono (usually preferred by FOH operators) whilst a stereo version is available via the TRS jack output, which allows to you retain the stereo element in the nice room simulations. Should you be unsure which virtual cab to commit to, there's a virtual DI box option to use instead of a cab, and that would allow you to use third-party IRs.

Whilst you are limited to saving three favourite virtual cab setups and three amp-channel presets to the hardware, there's no limit to how many of your own setups you can save in the Architect software. You'll also find a useful collection of Blackstar's own presets there that make great starting points while you get familiar with the software. And don't overlook the EQ presets; somebody with a very good ear has tuned those to really get the best out of the different virtual mic types.

The Last Word?

It's not uncommon for products designed to cover a lot of bases at the same time to fail to be exceptional at any one of them, but that's very much not the case here. The amp modelling of the high-gain sounds is as good as anything I've heard, and the power amp works better than any other Class-D I've experienced into a real speaker. It's quite difficult to know exactly what qualities the actual power-amp stage is bringing to the output when the modelled power-amp response, so convincing both in sound and feel, that you hear when you are using the CabRig output is still in play when you are using a real speaker. In the final analysis it doesn't matter: both real and virtual cabs work very well. Just don't try to monitor them both at the same time, as they aren't time-aligned. A real cab behind you on stage with a CabRig feed to FOH won't be a problem, but if you are recording both streams, make sure it is to separate tracks so you can align the waveforms. Not that I think you'd benefit much from using both. I quite often preferred the CabRig version to my own miking!

Whilst the amp modelling is generally very good, I found some of the effects slightly underwhelming. They are certainly adequate for live use, however, which is probably the only context in which they would be used, and there are always pedals and the effects loop for when you want something special or specific. Self-noise (as opposed to noise picked up by the guitar) is remarkably low considering the amount of gain on offer, and latency, at around 2.5ms input to output, is imperceptible to me. With so much gain on tap in the Amped 3's Overdrive channel, there is a fair bit of intermodulation distortion, but it is the sort you can 'play with' if you know how to use it — playing high-string fourths will give you an octave below the upper note, for example — and mostly just adds to the impression of having a thunderously loud quad box behind you.

Blackstar don't seem to have gone down the Universal Audio route of modelling every last quirk and detail of specific examples of amplifiers in their Amped range, but rather to have sought to create classic sounds and behaviours that any tube-amp fan would recognise and be able to work with. Personally, I appreciate that both approaches exist, and whilst I love my UA pedals for what they do, there are also times when the Blackstar approach wins out. Blackstar's Amped 2 and 3 models are sure to appeal to anyone looking for something equally at home on stage and in the studio, but the fact that they are still a serious contender for anyone looking to cover just one of those roles means the Blackstar team have done a great job here.

£ Amped 2 £539, Amped 3 £499.
Prices include VAT.

W www.blackstaramps.com

Pittsburgh Modular Taiga

Semi-modular Synthesizer

Think you know what analogue synthesis sounds like? Pittsburgh's Taiga might change that...

ROBIN VINCENT

I used to have Pittsburgh Modular pegged as a straightforward, solid and no-nonsense purveyor of quality modular. That seriousness culminated in the fascinating complexities of the Voltage Research Lab released around Superbooth 2019. However, there was a hint of strangeness, an offset, a playfulness that seemed to mark a shift in Pittsburgh's intentions. Over the pandemic, the solid straightforward modular seemed to fall away as founder Richard Nicol struck out on a new adventure of animal-themed modular that was both peculiar and evolutionary.

Nicol's thoughts and discoveries emerged as limited-run Eurorack modules and then pushed into the mainstream through contributions to Cre8audio products. Working with co-conspirators in the shape of instrument builder Michael Johnsen and inspired by the great outdoors, the strangeness has built until it had no choice but to spill out into a fully formed synthesizer. That synthesizer is called the Taiga.

You could say it sort of smashes together the Cre8audio East Beast and West Pest into a hybrid of East and West synthesis, but that doesn't feel like the whole story. Don't be fooled by the normality and familiarity of the front panel; underneath is a deep oddness and delightful weirdness that makes you question what you know about analogue synthesis.

The Taiga is a three-oscillator analogue monosynth, and my first thought was that this is Pittsburgh's Minimoog. Then the waveforms start to warp and fold in on themselves. The filter fights the mysterious Dynamics section for harmonic content as it plucks, drones and pumps resonance into a low-pass gate. While it has similarities in synthesis, it feels like a very different animal to the Moog.

Overview

The Taiga follows the usual left-to-right convention of signal flow. The three oscillators generate their waveforms on the left before meeting up with a noise generator in the mixer section. The mix gets funnelled into the filter before arriving at the Dynamics section. An analogue delay rounds out the journey before the sound emerges from the mono output on the far right.

For modulation, there's a single LFO and two ADSR envelopes, one normalled to the filter cutoff and another to the Dynamics level knob. Added extras include a patchable preamp circuit for overdrive, a Sample & Hold module, and a Digital Control module for MIDI manipulations. Everything is patchable and fully modular, making it a useful collection of Eurorack modules in 60HP if you prefer to see things that way. But it is prepatched and ready to go for instant synthesizer action.

To play it, the Taiga favours a MIDI keyboard, which you plug into the Digital Control section over on the far left. From there, the signal flow and modulation are all very familiar.

First Impressions

It feels like a good chunk of synthesizer as you first put it on your desk. It follows the 60HP desktop synth format of the Moog Mother-32, albeit with a lot more

colour, and you can pull it out of the case and drop it into your Eurorack. It feels solid and meaty with good movement to the knobs and a satisfying clunk to the insertion of patch cables. It's powered by a wall-wart which is the only connection on the back, and comes to life with a dazzle of orange LEDs.

Bypassing any manual-based guidance I dove straight in, and my first play felt like I was undertaking an experiment. As with any analogue modular synthesizer, the knobs are not necessarily going to be pointing to anything useful and initially, it took a bit of work to pull it together. I found myself stumbling about trying to tune the strangely ranged oscillators, and then flicking between Dynamics modes, CV depth and envelopes to try to discover how the sound works. The patchbay initially seems a bit cryptic and enormous, with 60-odd patch points. And then there's the glow from the orange LEDs that are just a bit too bright to the point that they obscure the front panel around them and make the selections difficult to see.

Fortunately, the manual is very comprehensive and the Taiga began to unfurl over time. At the time of writing the manual didn't have any patch examples, but Pittsburgh Modular are putting together a patch book to get over those first stumbling blocks so you can easily dial in an initial three-oscillator synth sound. And by golly, once you've got it, it's a nice three-oscillator synth sound.

Oscillators

The three analogue oscillators are identical and offer a lot of waveform shenanigans. Each begins with a saw-wave oscillator core and then the waveforms are generated via a series of cascading waveshapers. This interesting approach gives us a range of wave shapes and waves between the waves. Pittsburgh say that you shouldn't really see them as defined shapes and instead use the term 'Seed' to suggest that the waveforms are growing towards being the sine, triangle, sawtooth and square waves that are printed in the front panel. And this lets them bend the waves to the places in between, so you also get something along the lines of a sine/triangle, triangle/saw, saw/square and square/sine. So that's eight distinct waveforms to choose from, although some are more distinct than others. And you don't even have to choose because if you light all the LEDs, that turns on waveform-party mode, and you'll get a random one on each keypress. That's an awesome and enjoyably quirky feature. The standard waveforms have a few kinks, but if that lets us find these other weird and interesting between-wave timbres, then that's a price worth paying.

Once you've decided on your waveform Seed the oscillator delivers it to a wavefolder controlled by the big yellow knobs. It can fold the waveform up to six times, pulling in additional harmonics and complexity with each turn. These folds build on the already wide variety of waveforms on offer, giving you quite extraordinarily versatile sound sources. Don't forget you have three of these, all running unique waveforms that will get blended in the mixer.

The oscillator tuning takes a bit of getting used to. They behave differently depending on whether you have them coupled up to MIDI or not. You press Shift and the Seed button to couple each oscillator to MIDI. With MIDI engaged, osc 1 gives you just the fine-tuning of a couple of semitones over the whole range of the knob. Osc 2 gives you an octave and osc 3 gives you two octaves. This tripped me up on my first play because I couldn't work out how to get them in tune. There's only one spot at which the pitches align and that's straight up at the 12 o'clock position. Everywhere else produces interesting chords and ratios. After you've got the hang of it, it feels intentional and really quite clever.

Disengage the MIDI and you have three full-range oscillators that have loads of room at the bottom end for acting as additional LFOs. In fact, you only get to a decent usable pitch at about the two o'clock mark. This makes them excellent companions for a bit of FM. They all have FM inputs and a knob control that is usually wired to the main LFO. There's plenty of fun to be had with that.

The oscillators are all standalone VCO modules with CV inputs over pitch, FM, sync and shape. They have an output for the shaped and folded waveform plus a separate sine wave output, which is great for sync and FM. My only gripe is that the pitches of the oscillators are not normalled to each other. So, if you wanted to play the Taiga as a three-oscillator synth with a CV controller or sequencer you'd have to split the pitch CV and patch each oscillator separately. The Taiga has a two-channel splitter so you can double up on oscillators, but having a third seems like a no-brainer. I should also note that on my early production unit osc 1 doesn't track as well as the other two over CV and starts losing semitones after four octaves. The other two are fine and they all track together perfectly via MIDI.

Mixer, Noise & Preamp

Once you've finished fiddling around in the oscillators, they travel to a four-channel mixer, which Pittsburgh point out is actually a combined two-channel and three-channel mixer, although it doesn't look like one. On face value, you can control the level of each oscillator plus mix in some noise from channel 4. Twelve o'clock is best for a clean signal and it will distort if you wind it around further.

Each channel is available in the patchbay, so you can mix your own sources of audio or CV. You can also split it down to its 2+3-channel origins by patching the 1+2 output. This removes channels 1 and 2 from the mix and leaves 3, 4 and the preamp on the main output.

The preamp is a grungey little circuit that's only available through the patchbay. It gives you up to 30x gain and overdrive on anything patched to it. Patching the 1+2 output to it seems like a good idea, and indeed it is.

The noise channel is used by the Sample & Hold circuit that's also only

»

Pittsburgh Modular Taiga

£699

PROS

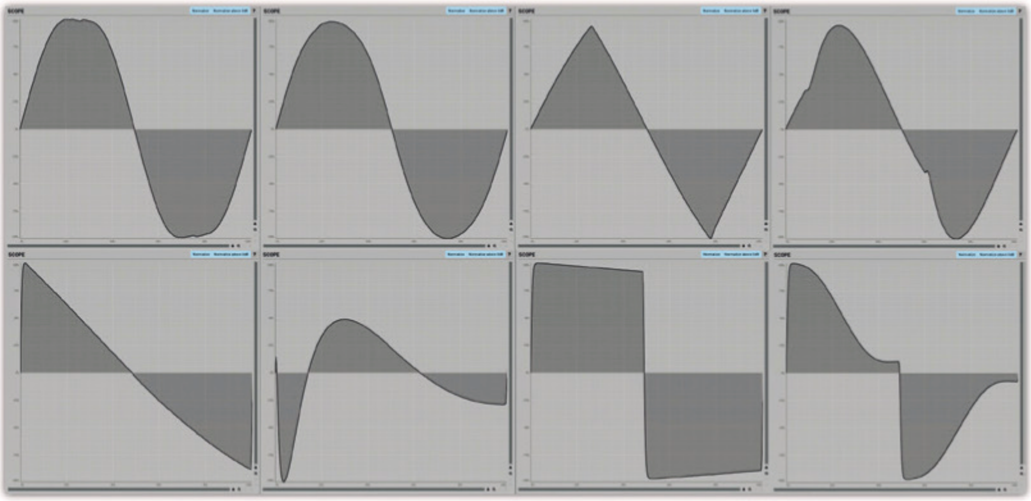
- Three analogue oscillators.
- Huge variation in waveforms and wavefolding.
- A filter that's all sweet spots.
- An organic and versatile dynamics section.
- Sounds unusual.
- Completely modular.

CONS

- Those bright LEDs.
- Can feel crowded.
- Paraphonic mode only produces two notes.
- Imperfect signal path.

SUMMARY

Taiga rolls together East and West Coast flavours into a heady mix of wavefolding, filtering and natural decay, for an unusual and powerful analogue synthesizer.



The eight waveforms available from the Taiga, some more normal than others...

available in the patchbay. It's normalled to both noise and the clock but alternative sources can be patched in.

PGH Filter

Pittsburgh are rightly proud of the PGH Filter and it's largely what made the East Beast so satisfying. It's based around a 2-pole 12dB state-variable filter and has no discernible dead spots. It doesn't matter how much you push the resonance; the sweep of the cutoff is smooth and consistent throughout. This makes it really good fun to play with. The low-pass is solid and gooey, the band-pass is clean and the high-pass sparkles.

As with the oscillators, Pittsburgh were not content with regular modes of operation, and so we find combinations of modes to give us the filters between the filters. This includes a rather sumptuous notch filter. Light all the LEDs and you will get another random mode of filter possibilities. Mix it up with random oscillator waveforms, and it's actually not as chaotic as you'd expect, just lots of room for movement, strange changes and discovery.

Paraphonic Mode

Pittsburgh describe the Taiga as a 'Paraphonic Analog Synthesizer', but I'm not convinced it's as paraphonically useful as that suggests. Through the magic of some correct settings in the Digital Section, the Taiga can manage two-note polyphony, which is then articulated paraphonically through the filter and Dynamics. You were probably hoping for three notes, but it's not to be. You can dial in chords with the three oscillators, but then you can only play them with a single key press. So, two notes it is.

This works by pulling the second note out of the Velocity output in the patchbay. You patch this to the pitch input of one of the oscillators that has been disengaged from MIDI. You can then play two notes together. It's a bit weird and not an altogether satisfying experience. It takes a while to set up, and longer to find something that sounds like it's not fighting with itself. I think Pittsburgh have slightly overplayed the paraphonic nature because it's not, for me at least, where all the fun is. Maybe it's more accurate to say that the Taiga is a great monophonic synthesizer with some paraphonic potential.

The resonance never quite pulls itself apart into self-oscillation and will, perhaps, be too musically well-behaved for some. You can get some overload and distortion in the peaks, but it's just on the finer side of fierce.

For modulation, the filter has two CV inputs with knobs for the cutoff. The LFO is normalled to the unipolar one, and the first envelope is normalled to the bipolar one. The LFO has two ranges; really slow to mid and then mid to really fast. This is not something I like very much because the area I want to play always seems to be sweeping through the switching point. Anyway, I can always patch in one of the VCOs to do that for me.

The power of three slightly detuned saw waves, pulling through an envelope-tugged low-pass filter that's sweet all the way around, is lovely and gives you a thoroughly satisfying synthesizer experience. But then, as you play with the shapes and start folding waveforms together or individually, it changes how you approach the filter. The harmonic content is different and modulating; it gives it a bit of a wavetable vibe which is both odd and very interesting.

Dynamics

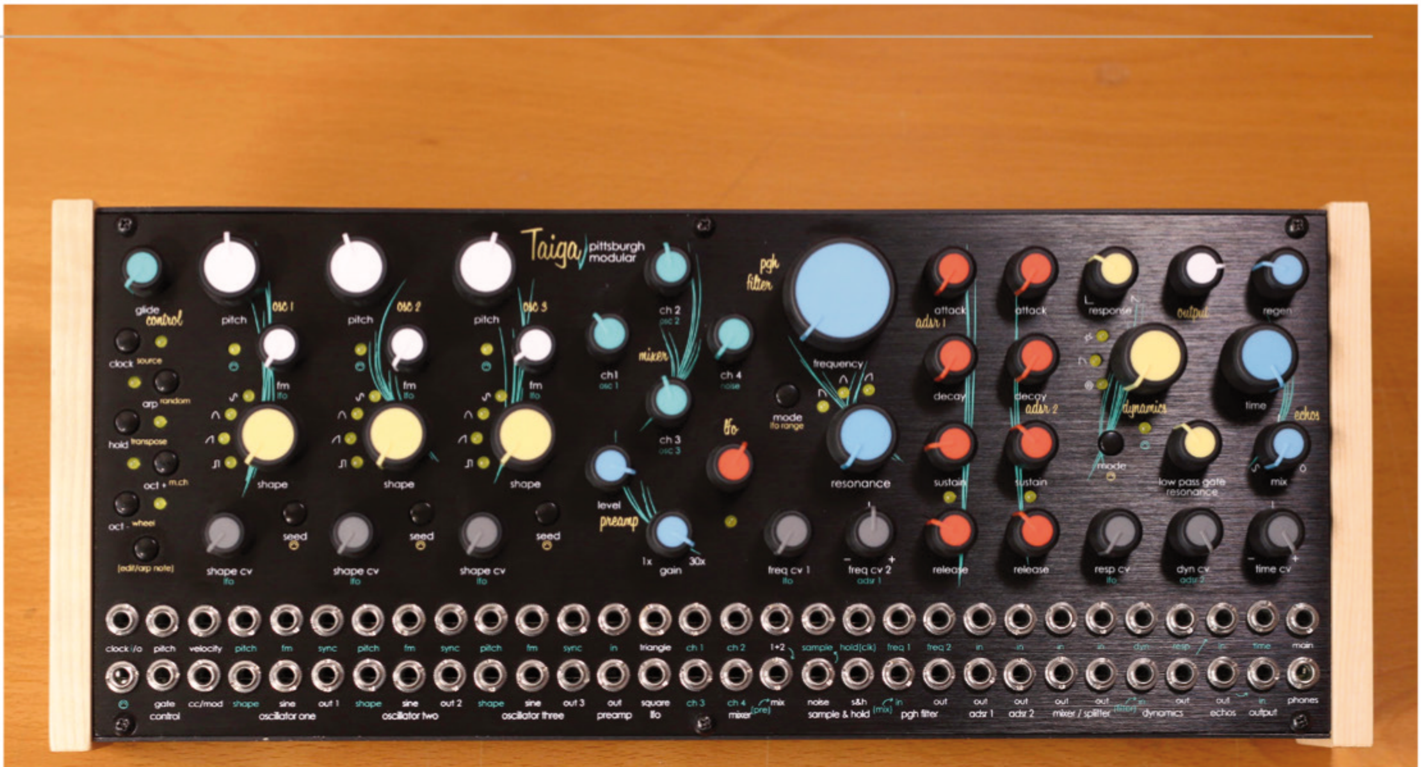
The other source of oddness and interest in the Taiga is the Dynamics section. This is where we find a unique combination of West Coast organic-ness, East Coast VCAs and unexpected harmonic filtering.

First of all, it has a regular VCA mode for regular things us Moog-style people understand. The Dynamics knob acts as VCA level and the Dyn CV knob acts as an attenuator on the incoming CV from ADSR 2. There's nothing unusual or interesting going on there.

Then we have two low-pass gate modes that bring a whole different feel to the Taiga. Low-pass gates (LPG) are usually seen as something from the West Coast style of synthesis. They use a voltage-controlled filter (VCF) to act like a VCA. So, as they close in response to decaying CV they filter harmonics rather than level. Vactrols are commonly used to control decay, giving them that short, plucky and organic feel that we associate with LPGs. The harmonic filtering and plucky decay all add up to a very natural and sometimes supernatural sound. Pittsburgh wanted to expand on these concepts to give more control and shaping, and that's what the Dynamics section in the Taiga offers.

In LPG mode, the Taiga uses a combined VCF and VCA to let you shape the dynamics with an entire ADSR envelope. There's an interesting interplay between the envelope, the Dynamics knob and the Dyn CV level where you can push it between sounding more like a filter or more like an LPG. With the Sustain set to zero we get full-on LPG. But rather than having the decay set by a vactrol we have a variable Response time set by the knob at the top. This introduces more decay than you usually find in an LPG while retaining all the harmonic characteristics.

The second LPG mode, Plucked Low Pass Gate, introduces a percussive spike at the front of the envelope. It gives it that familiar woody or tubey sound that's very Buchla-like and is present even with the Dyn CV turned down. It gets really interesting as you introduce a bit of the



All the Taiga's connections except the external power supply input are found on the 60HP front panel.

fuller signal back in, so you have this organic transient in front of the sound.

In both LPG modes, the filter resonance can be emphasised with the Low Pass Gate Resonance knob. In some respects, it makes the PGH Filter a bit redundant. Although, that's not right because the feel of the Filter and Dynamics systems are so very different. I found that I tended to lean on one or the other depending on what sort of sound I was going for, but it's fun exploring the slap that LPGs can bring to a heavily filtered bass line or the interplay between a modulated sweeping cutoff and the harmonic carving of the Dynamics.

There's something very earthy about the Dynamics section. It has a nuance to it where the sound is affected by subtle changes to parameters and small movements of envelope and response. You get a lovely roll-off when you strike or pluck a key very smartly that then disappears to something more solid as you play a bit more normally. The Sustain level on the envelope essentially places the cutoff point and dictates how bright the sound is going to be before falling away. If you're very familiar with subtractive synths, then this will feel nicely odd and strangely brilliant.

Echoes

At the end of the signal chain is an analogue bucket-brigade delay based upon a pair of 4096 stage BBD chips. There's not a lot of time on offer, and it's all a bit lo-fi. It can manage up to 250ms at full turn, although you can push it further by putting some voltage into the Time CV input at the expense of fidelity and some unpleasant high-frequency clock noise bleed.

You can get some nice and grungey feedback when you turn up the regeneration, and there's plenty of scope for some cool sound design, but it's not as musically interesting as I hoped. It can quickly get muddy and lose itself, but it does add another flavour.

Digital Section

On the East Beast and West Pest, the digital control system seemed a bit complex, with lots of button combinations and a cryptic display you were unlikely to remember. With the Taiga it feels much clearer and simpler, although you probably still want to keep the manual to hand.

There's a clearly labelled and easily accessible arpeggiator which simply arpeggiates the notes you play in the order you play them. A Hold button and Octave buttons come into play, making for a really nice experience. Some secondary Shift functions let you randomise the pattern or transpose while in Hold mode. Holding the Shift button lets you enter up to 32 notes in a sequence, building on the melody as it plays.

There are other functions hidden away in here, like a Pseudo-Random Shift Register and MIDI CC-controlled decay envelopes, random voltages and LFO. About half the manual is devoted to this section, and it's possible that I only see it as simpler because I've ignored most of it. But there are plenty of things you can explore here on a wet Sunday afternoon.

Burning Bright

Pittsburgh Modular have done something remarkable with the Taiga. They've come up with an analogue synthesizer that carves its own timbres and tones in a space where we think we've heard everything. It has three satisfying sections of intrigue and adventure: oscillators that bend and fold, a musical and juicy multi-mode filter, and a Dynamics section that plays with harmonics in a warm and organic way. It has its quirks and foibles, like the filter never quite managing to close and a bit of bleed through the LPG, but that's not uncommon in analogue circuitry. The forest of knobs can feel crowded at times, and accidental nudges happen quite often, but they've packed a lot of modular into 60HP of space.

The range of tones becomes almost ridiculous when all three oscillators are randomising their wave warps while being bent out of shape and sent through random modes of a filter only to be struck by a low-pass gate on their way out through a regenerative feedback loop. But the Taiga is a superb synth to spend time with, to explore ideas you hadn't considered before, and be rewarded for every subtle movement with a deeply interesting source of timbres. ■■■

£ £699 including VAT.
W www.pittsburghmodular.com

Mixing Atmos: David Kosten

David Kosten's first commercial Atmos project was an immersive mix of one of the greatest albums of all time. No pressure...

SAM INGLIS

With much to learn, and much to unlearn, most mix engineers would want to ease themselves into Atmos by taking on a few low-key projects. That didn't quite happen for David Kosten. Instead of an obscure indie EP or television soundtrack, he found himself charged with Atmosifying Mike Oldfield's 1973 masterpiece *Tubular Bells*.

Thanks to his work with artists like Keane and Bat For Lashes, as well as his own Faultline project, Kosten is a very well established producer, but he cheerfully admits to having no track record with Atmos. So how did he land such a high-profile, high-pressure debut gig? "I co-produced and mixed Steven Wilson's last record [*The Future Bites*], and I sat in on the Atmos mixing side of it. And that's really where my Atmos journey began. I was sitting at Dolby HQ, watching this record go from two channels to a whole bunch of channels and loving it. And thinking, 'Oh wow. Pretty much everything I've ever worked on would've benefited from working in this three-dimensional format.'

"I think every artist would know from working with me that I'm obsessed with talking about three dimensions, and front-to-back perspectives. So I always talked in those terms, but Steven Wilson was actually physically doing it. For a long time, he had been saying, 'David, you really should be making, at the very least, your own music in this format because it's so well suited to it.'"

Filling The Gap

As part of the team behind *The Future Bites*, Kosten was then nominated for a Grammy Award for immersive sound. Nevertheless, he understood that he had a lot of work to do to get up to speed as an Atmos mixer. "The next step was putting the speakers in, which I did about a year ago, then getting my head down to learn it. And there's a learning curve.

"It's not as complex, technically, as I actually thought it might be. I understood

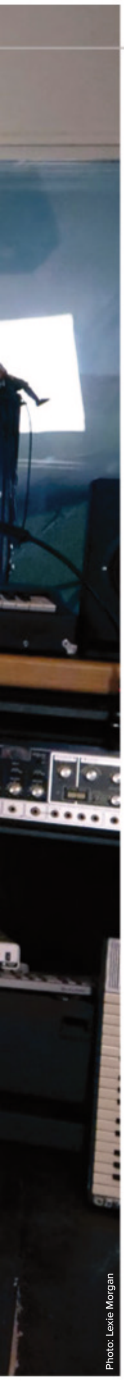
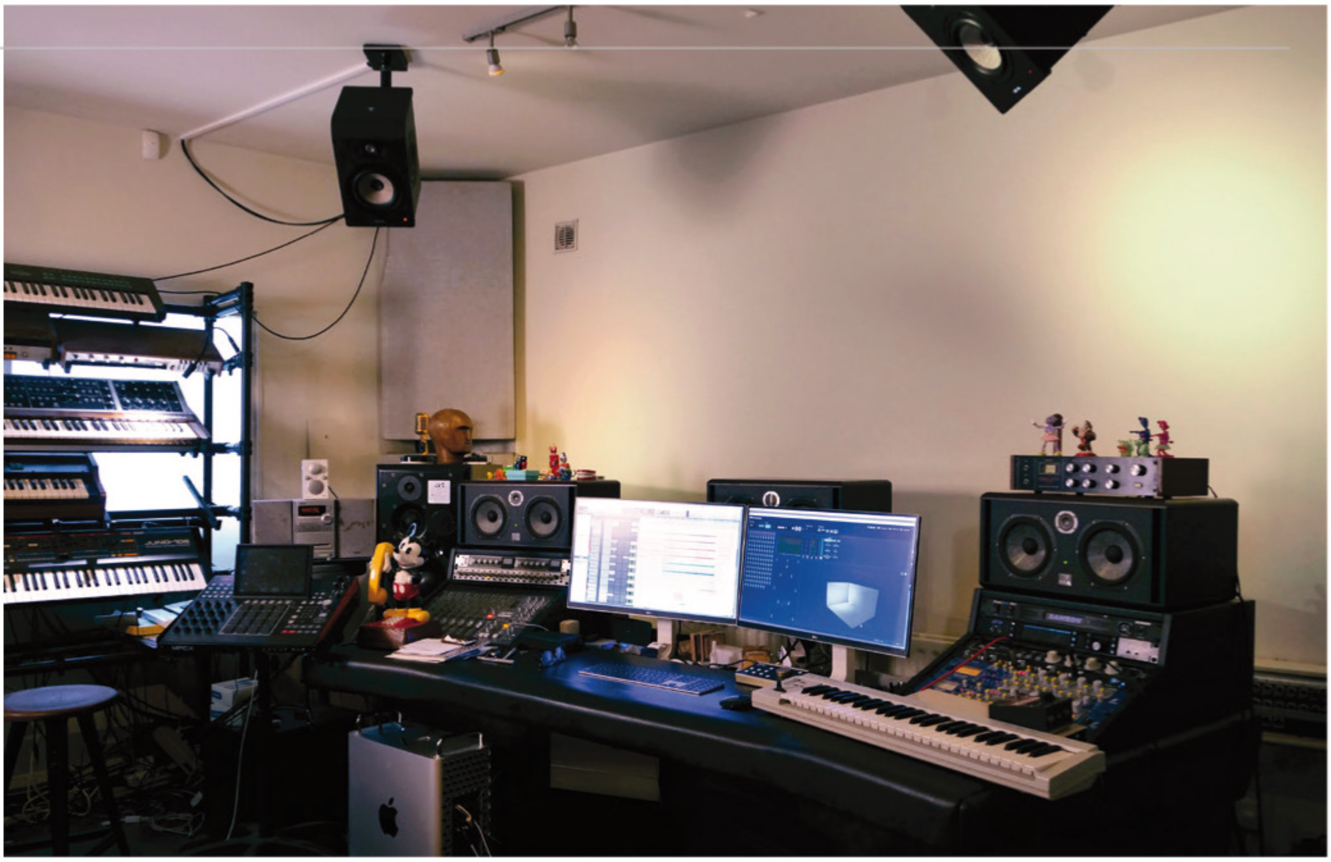


Photo: Lexie Morgan



To equip his West London studio for Atmos mixing, David Kosten supplemented his main ATC speakers with a Focal surround rig.

that pretty quickly, and I'd already had enough background from talking to Steven. But what I needed to do was work intensely on a lot of tracks, so I truly understood how to present music spatially, coherently and enjoyably, without it coming across on speakers as a collection of disparate sounds. And I worked on things that are already produced, went back to old mixes and converted them to Atmos, did some dummy runs with my own stuff. And Caroline Hilton (www.soundonsound.com/people/mixing-atmos-caroline-hilton) was unbelievably helpful and incredible with her time and knowledge. She really is the person who got me to this point where she would give me very honest feedback, and tell me when she didn't think something was working, and really helped a lot.

"It's pretty good being able to give your music to Steven and her and a few other people for them to critique and say, 'You know what you're doing now.' And so by that point, I felt like I was ready probably for a low-key first proper gig.

"So, then I got my low-key first proper gig!"

Back In Time

Although Mike Oldfield himself remixed *Tubular Bells* in 5.1 back in the early 2000s, everyone agreed that the starting point for the Atmos mix should be the original stereo album. Kosten's first and arguably biggest job, therefore, was to recreate the original stereo mixes as closely as possible, in order to generate stems that could be used for an Atmos presentation.

"My intention was that I wanted people to be hard pushed to tell that this wasn't done in 1973. It would just happen to have a time machine and be able to put it into spatial. I wanted it to sound exactly like the 1973 version. No excuses. There is no decision in terms of panning or EQ or effect or anything that I made. Every single decision that I heard on those stereos had to be duplicated as exactly as I could first. So I was essentially trying to mirror what the guys did in 1973 before I did anything to do with Atmos."

This, it turned out, was harder than it sounds, one reason being that important parts were simply missing from the surviving multitracks — among them the album's titular tubular bells. Although there were several bell recordings on the multitrack, none of them was the part used on the original album mix. Where did the bells on the album itself actually come from? Had they been overdubbed during mixdown? Or did Mike Oldfield erase them from the multitrack later? We'll probably never know, but it meant a lot of work for Kosten, who had to finesse that section of the stereo master into his multitrack.

"Weird things happen when you don't realise that your record is going to be an iconic classic!" says David. "Maybe you're a little bit more *laissez-faire* with masters, and there's no undo button on a two-inch 16-track. And he was literally filling every second of that tape.

"One of the things I was faced with early on, as well, was that there was no multitrack available for the end of Part Two, the Sailor's Hornpipe. So I had to use the 5.1 from 2009 for the end section because there was no multitrack from it. But then, obviously, that needed to be edited in perfectly to match the originals."

Exploding Tracks

Tubular Bells was recorded to 16-track, two-inch tape, and considerable ingenuity had been required to capture all of Mike Oldfield's musical ideas. Each track on the multitrack typically contained six or seven different parts, often featuring completely unrelated instruments. "Every second of every track has got something on it to get these ideas that he had out. So he's looking for space everywhere."

Since he was working in Pro Tools and thus not subject to the same limitations, Kosten split each musical element out to a separate track. The original 16-track recording thus ballooned to well over 100 tracks, most of which would eventually become stems to be placed within the Atmos sound field. But before that could happen, there were plenty more hurdles to be overcome.

Famously, each side of *Tubular Bells* is a continuous piece of music that lasts over 20 minutes, and is not broken up into separate songs. However, it turned out that the multitracks for each side actually

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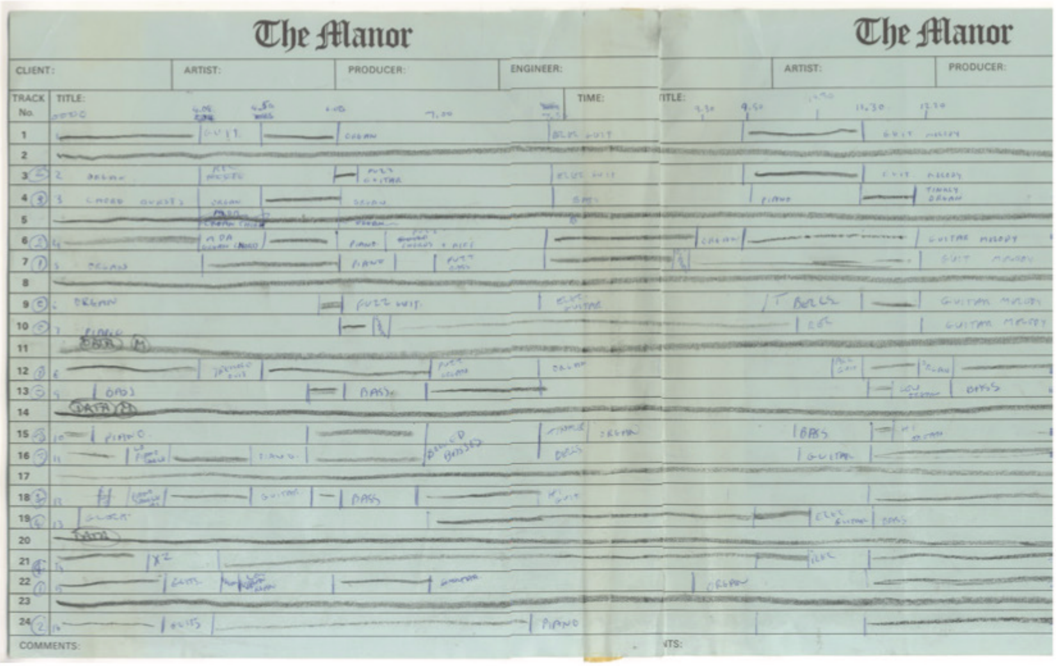


Although the original recordings were made to 16-track, the track sheets David Kosten was sent are for a 24-track machine. It's not clear whether the original engineers simply didn't have any 16-track sheets to hand, or whether these relate to a later 24-track transfer.

differed in length from the album as released, either due to speed variations on tape machines or edits made to the master tapes. Much painstaking listening and editing was needed to recreate the exact timing of the master.

"I had to do all those edits to then match the timing of the multitrack. So I used the structure of the original stereo and then I edited the multitrack to that, because I wanted to really be able to A/B between them. The timings of these things just never quite marry, they'll be tape machines running at a very slightly different speed and it's almost impossible to match."

Match & Mix

Once the basic arrangement had been recreated, and unused parts and studio chatter had been trimmed out, the next challenge was to rebuild the original mix. For an album recorded in 1973, you'd expect this to be fairly straightforward; engineers were in the habit of committing everything to tape during recording, so pushing the faders up on the console would usually deliver something recognisably similar to the finished album. Not so with *Tubular Bells*.

"I realised that, across getting on for 50 minutes of music, what I was actually listening to was the most astonishingly brave and uncompromising time spent in a recording studio that you could imagine. And the fact that it turned into this monstrously successful thing is mind-boggling really when you realise the extreme processing that was going on. What I had to do to duplicate those mixes was not gentle. They must have been just following their noses and thinking, 'I want it to sound like this. How do we do that?' By turning that knob — a lot!

"It's so uncompromising. And the first thing I realised as soon as I listened to those raw sounds was how in so many places the mix of the original record was extraordinary."

So radical was the EQ and other processing in places that it was hard to identify which sound on the multitrack was actually responsible for the end result. In one instance, for example, a full-range Hammond organ track had been high-pass filtered at the mix to the point where only a thin buzzing sound remained.

Never Standing Still

"What I think was also extraordinary," adds Kosten, "was how dynamic the mixing is. There's no point in the entire album where a sound is just left alone. The famous pianos more or less just sit there and do their thing, but then there's other layers of piano that come in and those have different processing: they'll have a different reverb, they'll be panned in a different spot. But it's dynamic in the sense that guitars will pan all over the place left to right and back again. Or they'll sit over on the right-hand side, but they'll get extreme wet/dry changes all the way through. And you hear it so clearly on the record.

"For instance, at the end of Part Two there are three or four different guitars, and you hear them totally wet, it's almost down a hole, down a ravine — and then it comes back and it's right next to you. All of that stuff had to be automated, listening on headphones, doing it a second or two at a time.

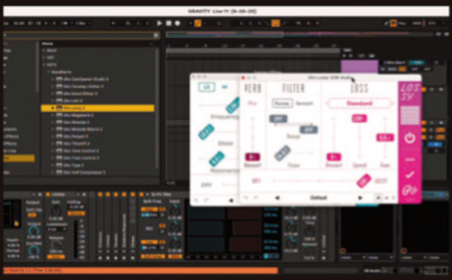
"There are also big organ stabs — I think he wanted them to sound like brass stabs, but he had a Hammond. And every single one had a different EQ and a different level. It was unbelievable how they evolve. A dozen or so stabs and they were all different.

"There is nothing conventional about any of this. I think at some point there's five or six bass guitars playing at the same time. You don't get that very often. So what does one do with that? You are just using your experience as a mixer to try and maintain separation in the same way that they did when they were making their stereos.

"There was also a decision to make as to what is good hiss and what is bad hiss. Good hiss, dirt, grime, mistakes: all of that is good if it's on the original multitrack and got used in the final mixes. But I didn't try to emulate the hiss of the original stereo quarter-inch tape. I thought that that would be a hiss too far. I did actually have to recreate hiss here and there and do funny things to try and seamlessly make sections work. There was one bit where they'd obviously used a processor that was very hissy, and it didn't sound exactly as the original did if I didn't add some hiss. I've got an amazing old Binson echo out there, which sounds fantastic but also has the right kind of hiss. So it's my go-to hiss creator. If you want something to transport you into a particular era, it's pretty amazing how well that works."

Source To Stem

Although the sources on the original multitrack were nearly all recorded dry



SOUND ON SOUND

VIDEO

VIDEO DOCUMENTARY ORIGINALS

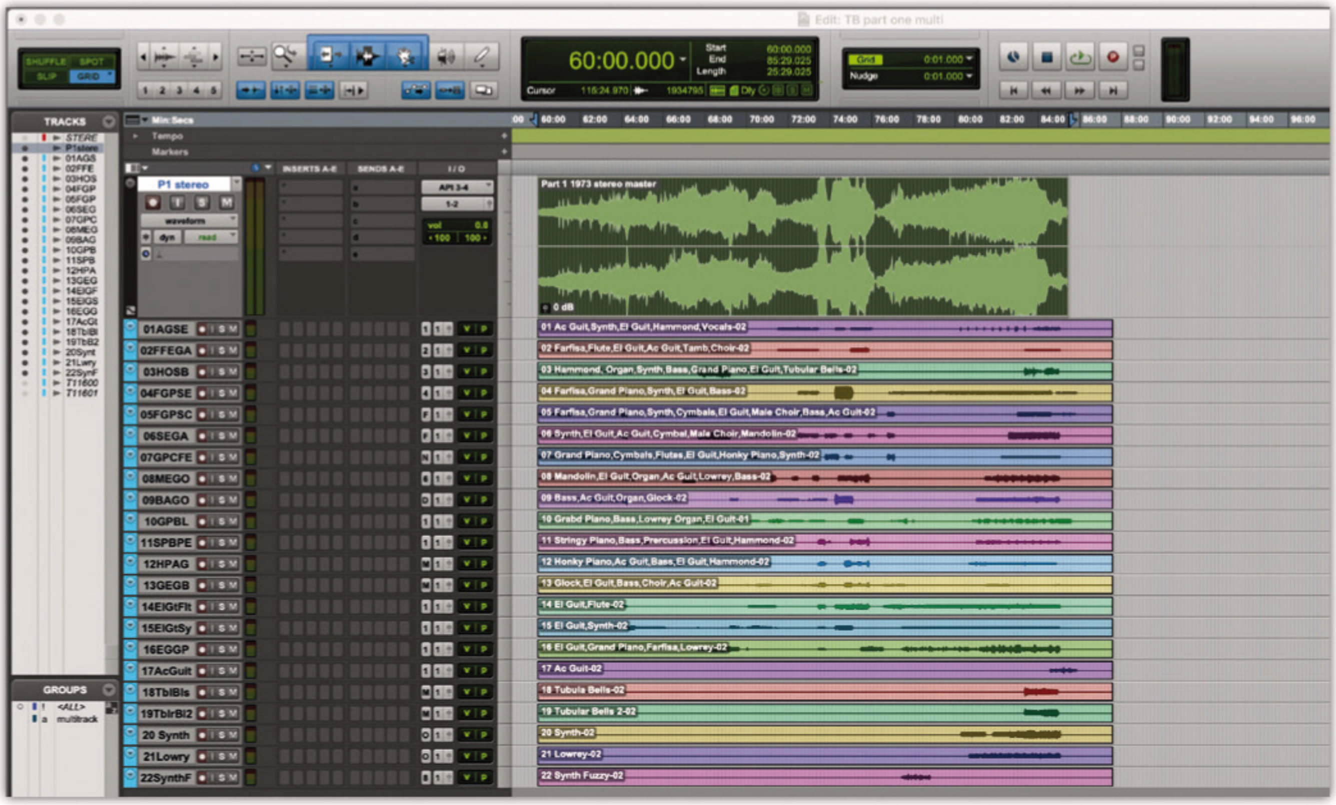
IN ASSOCIATION WITH

UNIVERSAL AUDIO

DJ DAHI: PRODUCING 'GRAVITY'

Californian producer DJ Dahi has worked with an incredible array of artists including SZA, Kendrick Lamar, Steve Lacy, Childish Gambino, Post Malone, Travis Scott and Dr. Dre. In our exclusive interview, Dahi talks about his production process and the value of working with real musicians, and breaks down his influential production of 'Gravity' for Brent Faiyaz featuring Tyler, The Creator.

www.youtube.com/soundonsoundvideo


Edit: TB part one multi
60:00.000
TRACKS
GROUPS
Part 1 1973 stereo master
0 dB
P1 stereo
01 Ac Guit,Synth,El Guit,Hammond,Vocals-02
02 Farfisa,Flute,El Guit,Ac Guit,Tamb,Choir-02
03 Hammond, Organ,Synth,Bass,Grand Piano,El Guit,Tubular Bells-02
04 Farfisa,Grand Piano,Synth,El Guit,Bass-02
05 Farfisa,Grand Piano,Synth,Cymbals,El Guit,Male Choir,Bass,Ac Guit-02
06 Synth,El Guit,Ac Guit,Cymbal,Male Choir,Mandolin-02
07 Grand Piano,Cymbals,Flutes,El Guit,Honky Piano,Synth-02
08 Mandolin,El Guit,Organ,Ac Guit,Lowrey,Bass-02
09 Bass,Ac Guit,Organ,Glock-02
10 Grabd Piano,Bass,Lowrey Organ,El Guit-01
11 Stringy Piano,Bass,Prercussion,El Guit,Hammond-02
12 Honky Piano,Ac Guit,Bass,El Guit,Hammond-02
13 Glock,El Guit,Bass,Choir,Ac Guit-02
14 El Guit,Flute-02
15 El Guit,Synth-02
16 El Guit,Grand Piano,Farfisa,Lowrey-02
17 Ac Guit-02
18 Tubula Bells-02
19 Tubular Bells 2-02
20 Synth-02
21 Lowrey-02
22 Synth Fuzzy-02

This Pro Tools session was the starting point for recreating the stereo mix of Part One. The original stereo master is at the top, followed by the 16 tracks from the original recording, and lastly some additional elements that were unearthed during the 2009 remix.

» and in mono, effects such as chorus, delay and plate reverb had been used copiously at mixdown. In David Kosten's recreated mix, therefore, the sources all yielded stereo stems with effects printed to them. Sometimes, in fact, he found it necessary to combine two or three different effects to recreate what was done on the original. These were often printed to separate stems in order to open up further options for Atmos panning. The aforementioned Hammond stabs were a case in point.

"There's three different reverb effects, plus all the EQs and things. I printed those separately and then moved the Hammond so you get your stab and then the decay of it floods from front to back. So if you listen to it side-on, you'll hear the reverbs shoot past you. So that was a bit of fun, but the actual sound of those Hammond organs needed all of those different effects to get as close as I could to the original."

Creating stereo rather than mono stems also meant that stereo motion panning — an important element of the original mix — could be baked into the stems. "By doing it that way, you're then maintaining the stereo imagery that Mike had, and then you can choose where to position it front and back. I like the simplicity of that system, rather than trying to mix it from the original multitrack and go straight into Atmos. It doesn't matter how you get there if you're creating something brand new. But this was so much about maintaining the integrity of those original sounds that this was the way.

"All the panning that I was doing was embedded in that stereo, so essentially, the Atmos part of it was more to do with... I feel like storytelling is the best way of describing it, even though it sounds horribly pretentious. That part of it was where I spent the time in the Atmos mixing. It wasn't in the panning because that had already been done. I made the odd little nudge of EQ here and there, but pretty much was happy with how the sound was printed. There's a little bit of automation that I did; for example, there's a piano that rushes towards you. If you listen to that on a surround system, you'll hear the piano start behind you and rush towards the front. So that was just drawing in some automation for that. But mostly all the panning choices were baked into those stems.

"I felt it was more important to just be able to know that the sounds were right, and then just allow it to breathe enough so it almost hovers around you. I didn't get into extremes, there's no fun with spinning sounds or anything like that. I've basically copied every single panning choice from Mike Oldfield.

"It was the placement of those stems that I spent a lot of time thinking about. If it's a strongly melodic element, or if it's a storytelling element that you want to focus on, that usually suggested that it was more in front of you than as a special effect behind you."

Objective Approach

Dolby Atmos is a hybrid format that combines one or more channel-based surround 'beds' with mono or stereo 'objects'. Objects are packaged with their own automation data, which is decoded on playback to determine their position and movement. Although he rarely used Atmos motion panning as an effect, Kosten chose to treat his stereo stems as objects, rather than routing them into the bed. "The control over the precision, the pinpoint ability to put it in a specific space, is much more what I wanted. And you just don't have that same control using a bed. Also, of course, for the binaural headphone mix, you can specify a different Atmos distance setting for every single stem as well. And by having them all separate, it meant that you could be much more selective about that.

HIGH FIDELITY
PURE AUDIO
TUBULAR BELLS
Mike Oldfield
SDE EXCLUSIVE
Blu-ray with Dolby Atmos, 5.1 & Quad mixes
50th Anniversary Edition

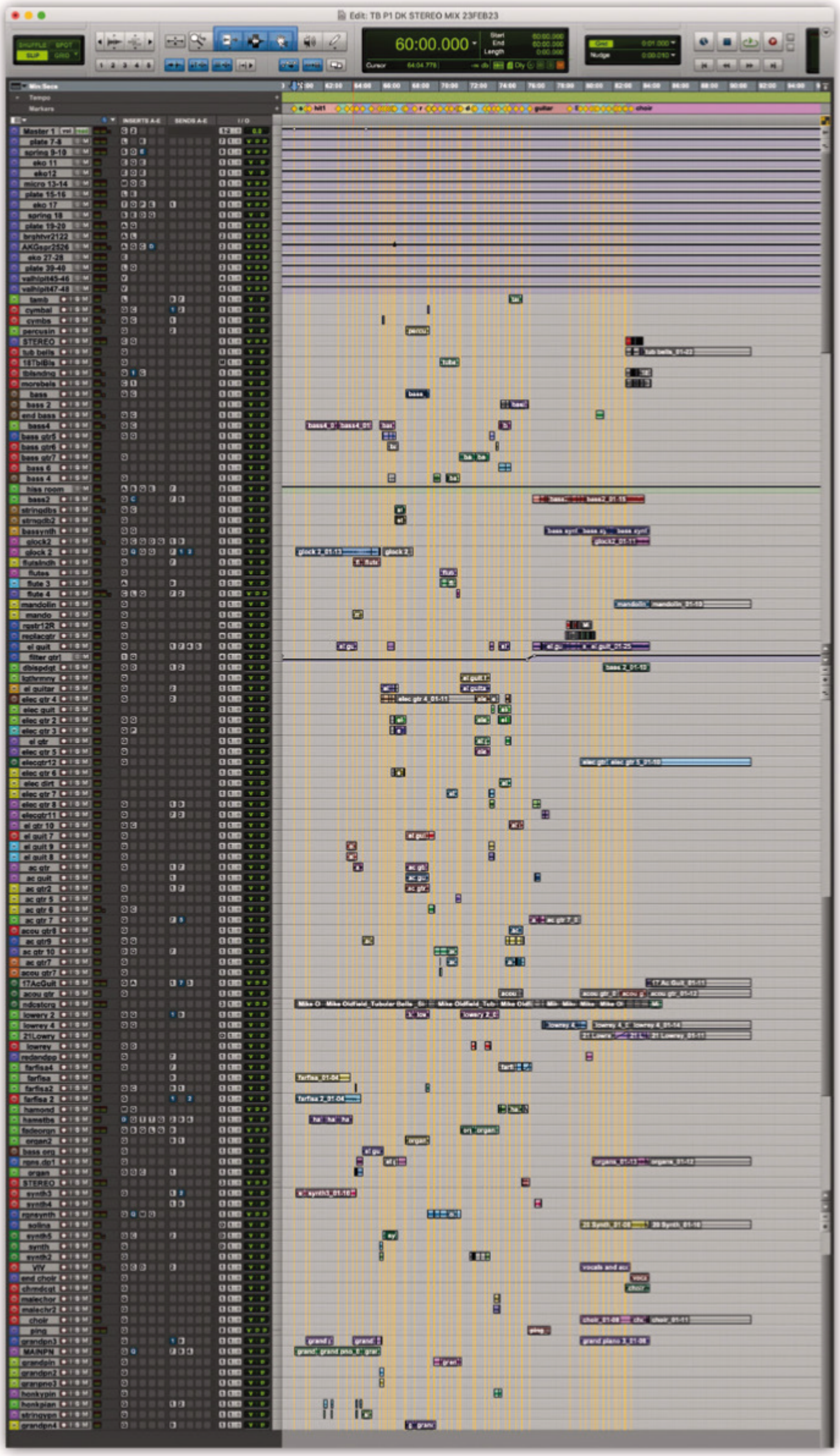
The 50th Anniversary reissue of *Tubular Bells* includes not only David Kosten's Atmos mix, but also his recreation of the stereo mix, alongside the original quad mix and Mike Oldfield's own 5.1 remix.

"I've heard awful binaural mixes where the lead vocals have the wrong setting, or they put too much reverb on their rebuilt stereo mix, or whatever. And I'm full of complaints about lots of Atmos mixes that I don't like. But by giving it a stereo mix that I was happy with, it felt fine. But obviously, I did the binaural mix from 80-odd stems, and it's a very unusual thing to have that control. Normally, we're presented with a tiny number of stems, and that's what most Atmos mixers are faced with. So they're having to do all sorts of things to try and make the Atmos mix sound reasonable. But here I was able to avoid that."

Integrity

Many engineers talk up the differences between stereo mixing and Atmos mixing, but David Kosten is emphatic that a correct stereo mix is the key to a good Atmos mix, especially when it comes to remixing a classic album like *Tubular Bells*. "I hope people can hear that it was done with the attention to detail that a fan would want. I'm pretty sure there's been rebuilds of tracks where that's not been the case, or fans have heard things and gone, 'Well, it doesn't sound much like the original. Why does it sound so different?' It's not to do with the Atmos part of it, it's to do with the agony. That's the amount of work that it takes to recreate what was going on in a studio 30, 40, 50 years ago. That's quite hard. So I have respect for anyone who tries it.

"This is all about honouring the choices that were made in 1973, but giving it access to a new, more immersive space. That was the whole point of it. So I didn't even want to get into the idea of creating something new for this piece. It was much more about how you maintain the integrity of that original stereo master, but gently bring it into 2023 without appalling anyone. And I'm sure it's going to have lots of people appalled, but my hope would be that anyone listening to the Atmos version of this would be able to hear the love and detail that I threw into it, because it was a hell of a labour of love to get to something that was even close to the original stereo." ■■■

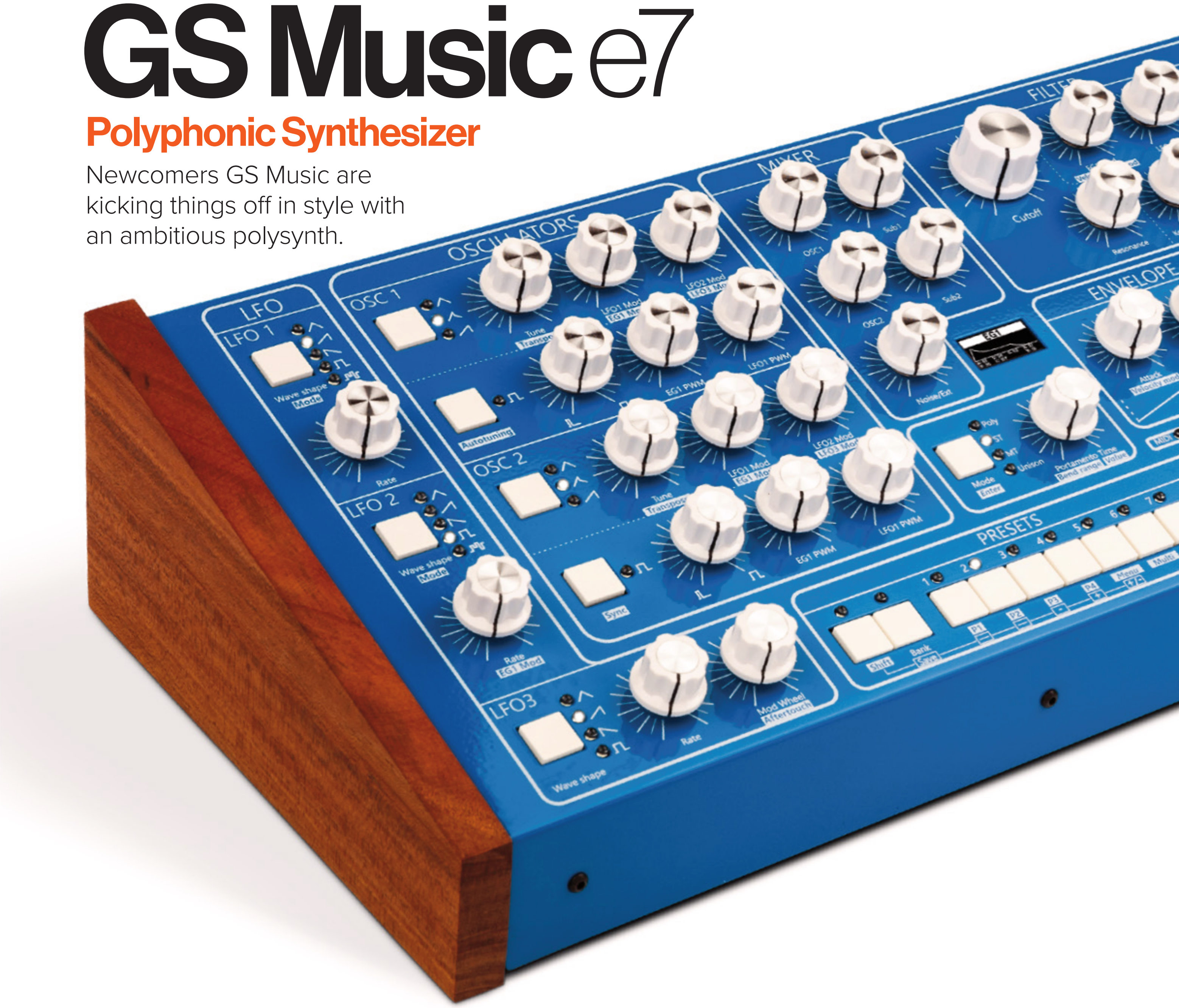


Once all the parts from the multitrack had been split out to individual Pro Tools tracks, the session ballooned to over 100 tracks in size.

GS Music e7

Polyphonic Synthesizer

Newcomers GS Music are kicking things off in style with an ambitious polysynth.



WILLIAM STOKES

Considering the GS Music e7 is purportedly the first ever analogue polysynth to be designed and manufactured in its developer's native Argentina, I'll assume it's no coincidence that not six months after the nation's momentous World Cup victory, a unit arrives at my studio sporting a striking white and light blue finish. This said, I should mention that the e7 is in fact also available in three other colour schemes: blue and grey, blue and black, and red and black, and all with rather nice wooden side panels to boot. It started life back in 2019 as a prototype called the Zeus, a 12-voice polyphonic keyboarded instrument not visually dissimilar to the Moog Voyager, before evolving into the seven-voice desktop synthesizer that sits before me today.

Out Of The Box

Suffice to say, unboxing a large-format desktop synth of the e7's ilk is always a nice experience. It's weighty, well-built and, well, it's big; 52cm by 19cm to be exact. I daresay that in today's world of sleek, scaled-down analogue instruments by boutique companies (just look at Superlative's Monolab System 1000, announced at this year's Superbooth), it's something of a statement of intent to make a palpably classic-looking desktop synthesizer; one more akin to the likes of Oberheim's desktop iteration of the OB-X8. The e7 presents a spacious panel, replete with Moog-style knurled knobs offering near one-to-one functionality (save the inclusion of a Shift key and a few simple menus). Its classic feel extends, for better

AMPLIFIER
OUTPUT
Master Volume
VOICES
ENVELOPE GENERATOR 2
GENERATOR 1
Attack
Decay
Sustain
Release
DELAY
CHORUS
Rate
Depth
Mix
Delay Time
Feedback
e7
GS MUSIC
Panel
Config

or for worse, to the slightly light, even plasticky feel of the aforementioned knobs, vaguely recalling vintage synths like the original Sequential Pro One. In fact, the e7 would look like it could be from any decade if it weren't for its little OLED screen.

I mean it when I say 'for better or for worse': while the build quality of the e7 errs on the raw side, it's worth saying straight off the bat that GS Music have done very well to keep the e7's price as low as it is. Don't forget that we still live in a world where many components are in short supply; I'd imagine a few creative workarounds were required to get the full bill of e7 parts to Buenos Aires and keep the whole operation solvent. I suspect its desktop form factor also helped in this respect, with GS Music ditching the Zeus's keyboard and prioritising connectivity over controllers (there's no onboard sequencer either). This it has glorious amounts of, with USB, 5-pin MIDI, CV and even MPE control all happily accepted by the e7.

The e7's rawness, I should say, is palpable: some of the physical knobs don't quite align exactly with their actual digital values, for example, and there's a faint but noticeable noise floor that makes me think of the Korg MS-20. This might have been an irritation if it wasn't for the e7's character and playability, which, like the MS-20, it has in spades. Its digital brain and rudimentary effects section make the most of an otherwise all-analogue signal path, boasting a variety of control modes and an impressive 600 preset slots. This said, the e7's architecture — at least on first glance — is based around a fairly conservative design. Or, should I say, an elementary one, in the sense that it also does well to present a great introduction for anyone into the craft of subtractive synthesis, helped along by its well-written and intentionally educational manual.

Playtime

The e7's two identical oscillators offer triangle, saw and triangle-saw waves, alongside a layerable pulse-width-variable square wave and additional square-wave sub-oscillator. They can be hard sync'ed, and there's also a handy autotune function, which snaps things into tune in an impressive two seconds or less. There's a noise generator which doubles as a volume control for external audio when connected, and adjustable portamento. So far so good. They're punchy, raw and feel ever-so-slightly drivable; not dissimilar to the incredible-sounding oscillators on the Analogue Solutions Leipzig, which I reviewed back in 2021 and still love to this day. In fact, it's worth pointing out that with no stabilising CPU on board the Leipzig, it was prone to some oscillator drift, which the e7, happily, is not. The e7's 24dB/octave filter is decent and smooth across its 10Hz-25kHz range, though its resonance never quite self-oscillates into true 'howl' territory, which felt somehow out of character with the rest of the e7's design. It has scope for plenty of movement beyond conventional enveloping, though, and can be modulated by any or all three LFOs simultaneously, as well as be influenced by the mod wheel, note velocity, aftertouch and keyboard tracking.

As you might expect, two ADSR envelopes control the movement of the filter and VCA respectively. In addition to this there are two rather lovely elements of expression on both envelopes, accessed via the Shift key: the first is a velocity modulation parameter on both the attack and release stages, which shortens both the harder a key is played. The second links keyboard tracking to the decay stage, shortening the decay time as you move up the keyboard. This, say GS Music, simulates the perceived shortening of high note decay times on acoustic instruments like pianos and guitars, and it does so effectively. There's also keyboard tracking on the overall VCA level, which when dialled in tastefully also helps achieve this natural feel. I have to say that, despite all of this, I would still like to have seen a sustain pedal input, but as it is there's a fine array of keyboard-based expression on offer for keyboardists of that persuasion.

The e7's screen is a well-proportioned inclusion, helpfully showing graphical readouts of EQ filtering and resonance, envelope curves, oscillator frequencies

»

GS Music e7

£1799

PROS

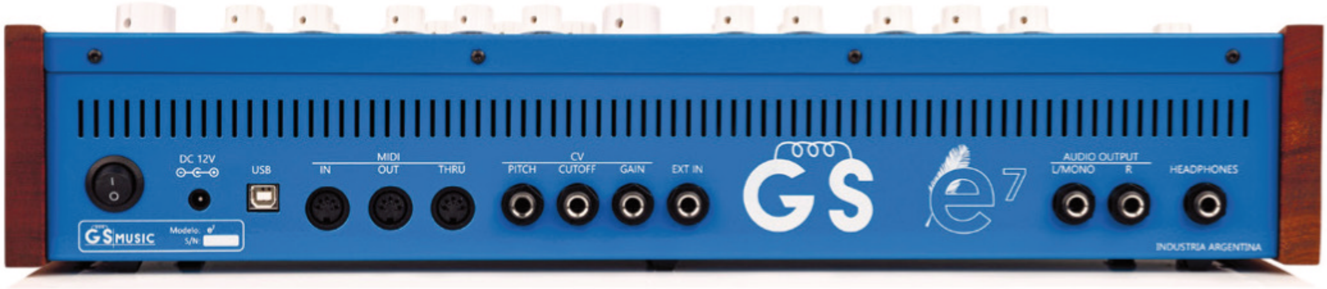
- **Great-sounding oscillators with multiple avenues of expression.**
- **Excellent Multitimbral mode with lots of editability.**
- **Highly connective across USB, 5-pin MIDI, CV and MPE.**
- **Great value for money.**

CONS

- **Slightly rough build quality.**
- **Could show slightly more aggression in some areas!**

SUMMARY

An impressive, characterful and reliable analogue synth with onboard effects and oodles of flexibility.



The rear panel offer full size MIDI in, out and thru, quarter-inch CV inputs for pitch, cutoff and gain, an external audio input and stereo and headphone outputs.

» and more. While the screen displays a moving readout of the chosen VCO waveform, don't be fooled into thinking this is an oscilloscope, as found on the likes of Korg's Monologue. It accurately tracks the pulse width of the square wave, yet does little else beyond showing what's already displayed on the physical panel next to the oscillator section. An oscilloscope isn't essential, of course, but it can certainly help with sound sculpting — particularly when tweaking, hard sync'ing, modulating and melding seven voices of analogue goodness. Perhaps that's for a future firmware update.

Enter The Modulation Matrix

Three LFOs furnish the e7's modulation matrix, each with slightly different functions. They can all generate triangle, saw, reverse saw and square waveforms, while LFOs 1 and 2 also offer random voltage generation. This I'm always happy to see included; more so two of them, and particularly in a polyphonic instrument geared toward functioning in a live setting, since randomness tends to be the very thing that can help make electronic instruments feel, well, live! There's a decent amount of flexibility here while still maintaining a relatively WYSIWYG interface: LFO 2's frequency can itself be modulated by one of the envelope generators, while LFO 3 is automatically set to be attenuated by mod wheel CCs, and can also be set to respond to aftertouch. Each can just about touch audio-rate frequencies, so while FM-style sounds don't come naturally here, they're not off the table.

The multi-mode nature of LFOs 1 and 2 is a real asset to the e7's sound: they can be set to oscillate in phase across all of its voices for huge, singularly undulating modulation; they can subtly move in and out of phase with each other across different voices for polyphonic depth; or they can retrigger a completely new wave for each voice, resulting in rippling, complex movements across chords and melodies. There's also a keyboard tracking mode that increases the LFO frequency with the note value, and clock sync'ing, with the option to have the phase difference between sequentially triggered LFOs quantise to clock divisions, which is smart. As someone who appreciates a visual cue where possible, I'd have appreciated an LED for each LFO to represent their respective movements in real time, particularly as the VCA envelope's behaviour is quite capably represented by the LEDS in the e7's Voices section, but as it is the LFOs don't leave much to desire. I daresay this would also have complicated things when it comes to triggering multiple LFOs in Polyphonic or Multitimbral mode, discussed anon. Over on the other side of the panel, in the VCA section, is a parameter for stereo spread, which when adjusted handily prompts the e7's screen to display where each voice is place in the stereo field. This can also be modulated for some simple and rather lovely Oberheim-esque stereo movement. Great stuff.

The e7's effects section is decent, presenting a chorus and digital delay. Similar to the e7's filter, the delay feels fairly safe, never reaching self-oscillation or any pitch-shifting chaos when shifting the delay time. As a means of bolstering the depth of the e7's output, though, it's very useful and does its job very well, as does the chorus. Both offer different modes; the delay can be clock sync'ed, usefully, and be set to unify across the stereo field or to ping-pong. The chorus has a Basic mode and an Ensemble mode, which sounds to me like it shoots an additional LFO through the phasing of the delayed chorus signals. The e7 manual proudly says Ensemble mode "adds a different dimension to the sound, simulating a string or vocal ensemble", and I'd say that's about accurate, if a slightly floral description.

Mono Power, Multi Madness

When it comes to its monophonic side, the e7 delivers equally well with copious amounts of variety on offer. Beyond the conventional Polyphonic mode, which retriggers both envelopes with each of the e7's seven voices, there's a monophonic Single Trigger mode (triggering the envelopes just once at the top of each held phrase) and a Multi Trigger mode (retriggering the envelopes with each note). There's also a single and multi-trigger Unison mode, lushly making use of all seven voices at once for some potentially huge sounds. While I could nitpick and say I'd like to have seen a single-trigger polyphonic mode thrown in there as well, I can't complain at this level of choice. Apart from anything else, I can safely say that if the e7 were exclusively a monosynth I would consider it a very powerful one, and that's before we even arrive at its four-part Multimbral mode.

Multitimbral mode opens up considerable sound-design opportunities in the e7 — both tonal and atonal — with the addition of moveable keyboard zones, which expands the scope of the instrument considerably. I enjoyed flicking through some of the e7's multitimbral factory presets, particularly one that used a combination of modulated noise and tonal oscillators to achieve a pulsing electronic cymbal and bass combination. When it comes to creating your own voices, it's a case of using the Preset buttons at the bottom of the panel to move between the four parts (or timbres), then sculpting a different sound for each: the oscillator, filtering, modulation, enveloping and more can all be individually set for each part to achieve anything from subtle, finely-tuned variance to wild and complex sounds packed with texture and different layers of movement. As with many other aspects of the e7, there's a healthy amount of limitation here: four-part multitimbrality might not sound as impressive as some other synths out there, but in the real world it's plenty for creating brilliant sounds. There's even a nifty on-screen graphic mixer to help strike just the right level balance between parts.

I should say that Multitimbral mode doesn't present the most intuitive way of navigating between different parts and editing them, with its combination of buttons and the screen menu, and on more than one occasion I found myself accidentally stumbling into a factory preset with no immediate way of returning quickly to what I was working on. But, as is often the case, things got easier with repetition. The only global values Multitimbral mode is subject to are the e7's effects and stereo spread settings, which if nothing else ends up a great way to keep everything cohesive, even at subtle settings. I almost found myself wishing for a one-knob compressor to cap it all off.

Having sculpted my four-part multitimbral ensemble I can say that while it was certainly enjoyable to have the sound lurch as one weighty block between notes on the keyboard or sequencer (indeed, sequencing this kind of thing at speed is always a thrill ride), the e7's moveable keyboard zones do a fantastic job of endowing it with true, musical applicability. They were also a breeze to program, courtesy of a screen-based readout of the chromatic keyboard and a visual representations of where zones begin, end and overlap. Strengthen the left-hand parts with some sub bass, add airy noise to the upper reaches of the keyboard, or have the two halves of the keyboard completely distinct from one another as if playing two different synths. It's also possible to have sounds blend and overlap with each other in one long continuum across the keyboard, adding all manner of exciting dynamism and texture to sequences and melodies as they move across the e7's range. It's even possible to have 'dead zones' in the keyboard, which could come in very handy if, for example, you're sending your controller to one or more other destinations alongside the e7 and want to keep them distinct from each other.

Other colours are available...

Conclusion

As I say, the e7 occasionally feels rough and ready, but its focus is demonstratively on the things that matter: sound and character. It's got depth and real substance, but it's also got attitude and edge. All things considered, it seems the e7 is seizing its moment to muscle in among the polyphonic analogue heavy hitters, more than happy to announce itself as a classic-in-waiting while punching above its price point quite considerably. Only time will tell if that wait will pay off in the public sphere, but it has all the makings of a mainstay on the synthscape and throws down the gauntlet to synths double its price. As an Englishman, I can only hope that when football finally does come home, we'll see a few home-grown synths hitting the market resplendent in red and white. If they're anything like this one, there'll be more than one reason to celebrate.

£ £1799 including VAT.
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SAMPLE LIBRARIES

Realitone

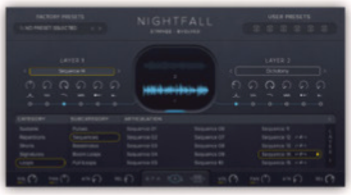
Nightfall

Kontakt Instrument

★★★★★

Billed as a hybrid/organic strings toolkit, Nightfall by Realitone is a follow-up to the company's excellent Sunset Strings. While it takes a darker turn from its predecessor, it still offers users the opportunity to create musically satisfying hybrid instruments between organic and more synthesized textures via crossfading or mixing. The potential timbral results are almost endless, and the instrument ultimately provides a deep tapestry to expand the capabilities of your existing libraries.

As in Sunset Strings, Nightfall does not contain legato performances, but rather a large collection of textures, loops, performances, sustains and aleatoric patches to combine and explore. One can imagine a scenario, for instance, where your existing string package is used as a legato instrument. Now is there a particular sustain which — after fading to a whisper — you'd like to further melt into a flautando, and then bend a whole step up as it disappears into a granular cloud? This is where Nightfall excels. Exploring further, I set layer 1 to 'Landscapes', a gorgeous and buttery texture built on what sounds like an orchestration of flautandos, muted strings and midrange harmonics, and which includes some organic but minimal movement, while layer 2 was set to 'Random Bowings'. As I used the mod wheel to crossfade evenly between the two sound settings, the gentle wash of the first texture gradually morphed into movement, and then into an aleatoric wonderland: it was great fun to say the least, and crafted an astounding 'nature emerging' type of patch. While the mod wheel is used to control gradual crossfades by default, it can be set to partial crossfades, which mix the two layers together, or natural crossfades. For those looking for hands-free insta-success, Nightfall also has an 'automatic mod wheel' feature where any setting assigned to the mod wheel ramps gently from top to bottom, creating automatically evolving sounds and leaving both hands free to play. In the above-mentioned combination, the mix of automatically changing filters allowed me to achieve everything from a modernist orchestra sound to an evocative dystopian landscape with just enough timbral variation to keep things endlessly evolving.



Bearing a similarly simple and intuitive interface to Sunset Strings, Nightfall has a separate effects area for each of the two layers. The controls include a 'tone shaper', a granulator, a low-pass filter, an LFO-like 'motion' filter, a delay and a lush onboard reverb. With controls automated or set to auto-motion in the hands-free mode, the timbral variation can be quite intense. If there is one complaint about this beautiful instrument, it is that it is hard to dial back the reverb at times, making the sounds very wet by default. Aside from this, there is very little to dislike about Realitone's Nightfall. Attractively priced, the package comes highly recommended! *Mark Nowakowski*

$199

www.realitone.com

Zero-G

Elements Cinematic Rhythms

Kontakt Instrument

★★★★★

ELEMENTS
CINEMATIC RHYTHMS

Produced by sample library legend Stefano Maccarelli and Zero-G, Elements Cinematic Rhythms is presented with a tagline of "contemporary tribal and warrior percussion". As we will see in a moment, that accurately reflects the style of the sounds. The library is designed for the full version of Kontakt (version 6.7.1 or above), offers some 15GB of sample data, and combines over 100 single instruments (with multiple dynamics layers and round-robin hits), hundreds of loops, an impressive collection of multi-presets, internal sequencing, close and room mic options and a range of effects.

The Kontakt front end may appear minimalist but, via the Expert button (which options a whole series of sliders to customise the sounds) and the Sequencer and FX Rack tabs, there are plenty of sound-design options to explore. The multitude of supplied presets are divided into two categories — percussion and sound design — and, usefully, the main UI is colour-coded yellow or red to reflect this. The former group presents the sounds in their natural form, while the latter group offers more of the hybrid sound, giving the user musical options depending upon the scoring context.

In terms of the sound sources, the instruments include bass drums, gran cassa, frame drums, various toms, djembe of different sizes, 'Warrior' and 'Village' drums (labels that are perhaps indicative of size?) and a whole range of smaller percussion elements including metal, wood, shakers, rattles, maracas, tambourines, cymbals, gongs and cowbell. Without exception, they all sound beautifully recorded and edited. What's noticeable is how natural they sound. For a 'big drums' virtual instrument, this is quite a refreshing (and useful) option compared to the more bombastic approach you might get in a library aimed at, for example, trailer music applications. Yes, you can create some forceful and attention-grabbing rhythms here, but the sound is not hyped and has room to breathe. The processed sounds offer a sonic palette with a more contemporary edge. Both are very usable.

The keyboard mapping of the samples — with loops, single hits and a selection of rolls arranged in colour-coded sections across the Kontakt MIDI keyboard — makes it very easy to build a complete performance. The loops themselves are really good and can easily be sequenced or layered to good effect. The additional icing on the performance cake, however, is the Sequencer option. This lets you create your own rhythmic patterns (with up to 32 steps) to trigger the sounds and, rather wonderfully, ships with an impressive set of pattern presets to explore. For busy composers needing inspiration to hit deadlines, there is plenty of help to be found here.

So, Elements Cinematic Rhythms sounds absolutely great, offers a broad selection of percussive sounds, and features all sorts of performance inspiration via the loops and sequencing features. Any media composer looking for some natural-sounding (not over-hyped) cinematic percussion could make good use of these sounds. And, given the surprisingly accessible pricing, almost any media composer will also be able to afford it. Elements Cinematic Rhythms is top-notch stuff and a real bargain. *John Walden*

£64.63

www.zero-g.co.uk

Fracture Sounds

Hemisphere

Kontakt Instrument

★★★★★

FRACTURE SOUNDS
HEMISPHERE
AMBIENT CINEMATIC GUITARS

I'll say it right now — Hemisphere is simply gorgeous. This ambient cinematic instrument adopts a similar approach to the electric guitar as Fracture Sounds' 'character' pianos Woodchester, Midnight Grand and Glacier Keys. Following a similar format, and using the 'house' GUI familiar to owners of the pianos, Hemisphere substantially elaborates on the concept with a varied range of carefully curated sound sources, additional effects, and some unique features.

As with the pianos, the raw guitar tone can be blended with three different Atmosphere pads which act as sympathetic extensions of the guitar tone. Hemisphere provides a choice of 15 different guitar tones, based on either a Stratocaster or a Telecaster, each sampled in different ways through various amps and effects. They range from big and dark to crystalline and heavenly, even gritty, but always with a gentle touch.

The format is also augmented with a fine selection of 14 contrasting Atmospheres, any three of which can be assigned to the mixer slots in any combination. A Transient Emphasis control applies shaping exclusively to the guitar tone; positive values emphasise the attack whilst negative values range from subtle softening to a long fade-in.

A Lo-Fi control offers much potential for sound design, comprising tape simulation with Age, Saturation and Warble, three types of noise, and a choice of eight speaker emulations; this effect is global across the whole instrument, the rotary control governing overall effect intensity. The tempo-sync'ed Echo includes saturation, damping and feedback, whilst the reverb comes with options for Room, Hall, Plate and the rather delicious 'Shimmer'. Also featured is a remarkable Fade-In parameter for the reverb, which I've not encountered before. Rather than being a standard pre-delay, the effect is introduced gradually for each individual note, thus reducing reverb pile-up and preserving the clarity of each note's transient.

Pitch-bending also features another innovation, whereby only keys being held down are affected; notes held by the sustain pedal are unaffected, allowing for an almost pedal-steel playing technique. Fret noise can be triggered either manually by keyswitch, or automatically (with variable chance), in which case it's only heard when all keys have been released — now that's great attention to detail. Other refinements include low- and high-cut filters, Timbre Shift and an Air control for the guitar tone, Colour (EQ), velocity response and sample start time, and the option to let the Atmospheres decay naturally or sustain constantly for as long as notes are held.

It's impossible not to be enchanted by Hemisphere. The 55 snapshot presets are inspirational, and when you want to get creative there are many ways to tailor your own sounds. This incredibly absorbing, immersive instrument sounds fabulous right out the box, surely destined to become an indispensable part of many a composer's toolkit. Just load it, play it, and fall in love...

Nick Magnus

$99

www.fracturesounds.com

Cinematique Instruments

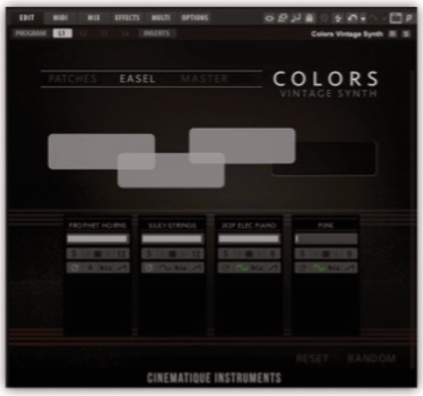
Colors Vintage Synth

Multi-format

★★★★

If you have a liking for the unconventional in your virtual instruments, then Cinematique Instruments are probably already on your radar. Colors Vintage Synth certainly ticks that box via its unusual UI. Sonically, it provides a source of pad, texture and ambience sounds most obviously aimed at media composers. For the purposes of this review, I had access to the version designed for use within Steinberg's Halion Sonic player (which is a free plug-in capable of running in any suitable DAW host), but there is also a Kontakt-formatted version available if that happens to be your preference.

The underlying approach is familiar enough in that you can build your own unique pads/textures by layering up to four individual sounds. Three of these can be drawn from a collection of 30+ vintage synth sounds (the total library clocks in at just under 1GB) while the final layer offers a small number of 'noise' sounds (including a very interesting Theremin option). The specific sounds chosen have obviously been carefully selected with the pad/soundscape/texture target in mind but they are drawn from suitably vintage origins including the Jupiter-4, Juno-60, Prophet-5 and Minimoog, among others. CI provide plenty of presets to get you started (and a Random button if you just want to roll the sonic dice) but you can easily configure your own 'sonic colo(u)r' combinations via pop-up menus available for each slot within the Easel page.



The Easel page's somewhat quirky control set is also where things get somewhat off the beaten track. As well as volume/pan and solo options, you can also adjust a simple arp, assign volume to the mod wheel, tweak the attack, change the tone via a simple EQ and apply a low-cut filter. However, in terms of adding a sense of movement to the sounds, the LFO is perhaps the star of the show. The Master tab then adds some further options. These include a global decay control, ambience and chorus, but the drive and filter features are particularly impressive. Again, in most cases, mod-wheel connectivity is available as well as further sub-controls. Yes, the implementation is a little unusual, but it's pretty easy to get your head around even for those who don't consider themselves synth programmers.

Colors Vintage Synth is very much designed with a specific musical function in mind — pads, textures and soundscapes — and the majority of the presets demonstrate that to good effect. They also show off the modulation options, so keep a finger on that mod wheel as you audition. It's not all pads though, and a few presets make excellent use of the arpeggiator function for a little melodic flavouring. And while it is perhaps not the broadest of underlying sonic palettes, in terms of capturing that vintage synth vibe — warm, full, rich — Colors Vintage Synth absolutely hits the spot. A modestly priced means for anyone to place some authentic vintage synth sounds into their own musical projects.

John Walden

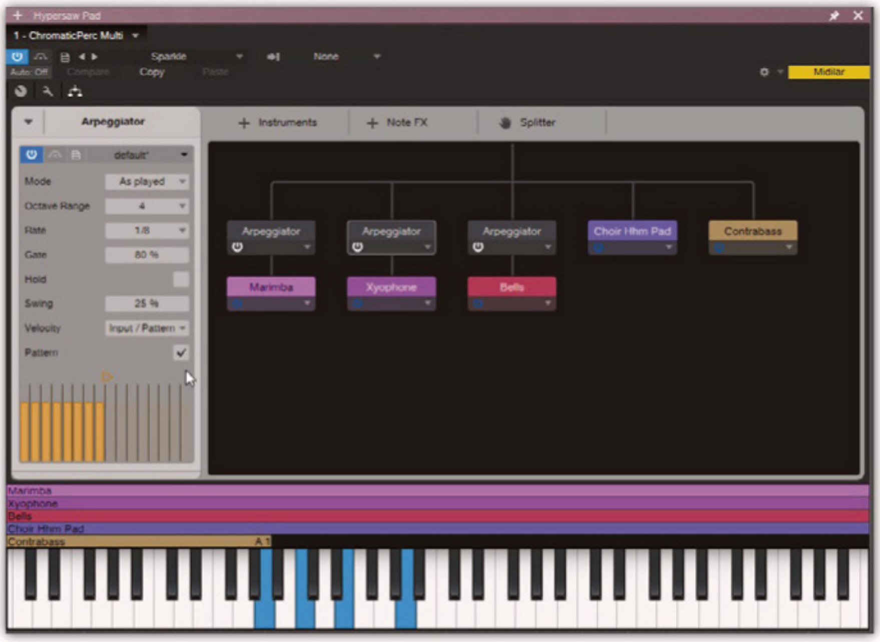
£51

www.cinematique-instruments.com

[Audio examples of this month's libraries are available at www.soundonsound.com.]

Multi Instruments are your fast track to powerful synth layers and splits.

ROBIN VINCENT



Sparkle
None
default*
Mode: As played
Octave Range: 4
Rate: 1/8
Gate: 80 %
Hold: ☐
Swing: 25 %
Velocity: Input / Pattern
Pattern: ☑

Screen 2: Press the Routing button and you'll see all the constituent parts of the Multi Instrument.

This month, we're going to take a look at the Multi Instrument, which can be an amazingly creative space within Studio One. In building our instrument, we will be looking at combining and layering synthesizers, adding effects, and how we can use automation to modulate them in a modular way.

The Presets

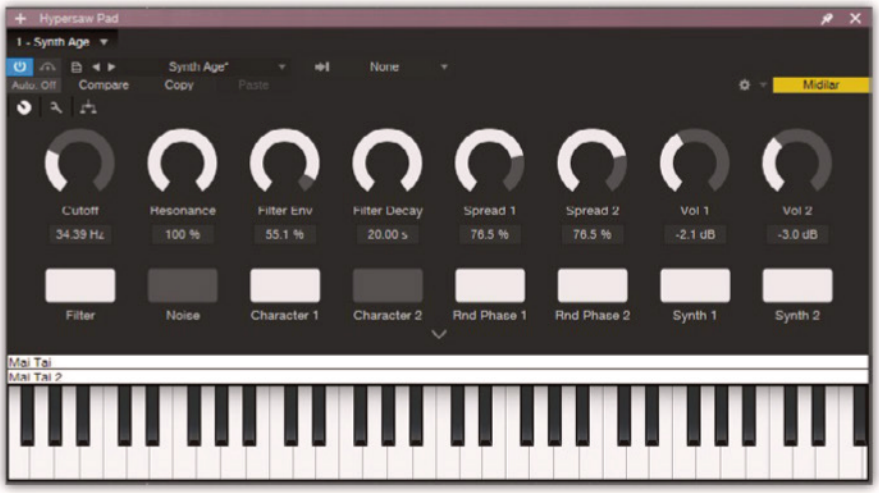
The Multi Instrument is available from the Instrument Browser, and it combines two or more virtual instruments into one. It can take relatively simple synthesizers and build them up into extravagant multi-faceted sound machines. You can combine a sampled piano with synth strings, add a guitar edge to a synth line, fatten up oscillators, or come up with massive environmental movements of sound using a single key.

Studio One comes with a load of Multi Instrument presets that use PreSonus' own-brand virtual instruments, such as Mai Tai and Presence. There are some excellent workstation-style sounds in here that none of the individual instruments could have pulled off on their own. Have a look at Suitcase&DynamicPad and Synth Age in the Layer section, try Sparkle in the Sequence section, or something beautifully simple like Herbies Split, from the Split section.

Multi Instrument Window

When you load one up, all you get is a simple row of eight knobs and eight buttons. These are macro controls that can be mapped to any of the parameters within any of the loaded instruments. If you push the little down arrow at the bottom, you'll reveal a pair of X/Y pads that you can also map to anything you want. You can combine parameters to a single control to control the filter cutoff on multiple synths all from one knob, for example.

If you press the Routing button (it's next to the spanner and looks a bit like a tiny attack robot) you'll get to see the contents of the Multi Instrument. If you load a preset like Space Nylon (in the Layer section), you'll see you've got three instruments loaded: Presence, with a Nylon Guitar, and two Mai Tai synths, one of which has a Repeater Note FX attached to it. Open up the Sequence / Sparkle preset and you'll find Marimba, Xylophone and Bells with their own arpeggiators, plus a Choir and Contrabass.



Synth Age*
None
Cutoff: 34.39 Hz
Resonance: 100 %
Filter Env: 55.1 %
Filter Decay: 20.00 s
Spread 1: 76.5 %
Spread 2: 76.5 %
Vol 1: -2.1 dB
Vol 2: -3.0 dB

Screen 1: When you first load a Multi Instrument preset, you'll see this simplified GUI with its eight macro controls and buttons.

You've probably noticed the keyboard at the bottom of the Multi Instrument window. The different loaded instruments are stretched across it, represented as bars of colour. This is where you can set the range for each instrument and create layers and splits to your heart's content. Any preset from the Split section will highlight what that's all about.

Effects

You can add effects to all the instruments individually, giving you a lot of excitement and movement. At the time of writing, the effects side is separate from the Multi Instrument. It seems to me that a natural evolution of the Multi Instrument would be to let you drop effect plug-ins right onto the audio outputs of those synths in the routing window, but you can't do that yet. Instead, you use the channel strip for the selected instrument within the Multi Instrument. Here, you can insert plug-ins and use sends, and adjust the level and panning as you would in any channel strip.

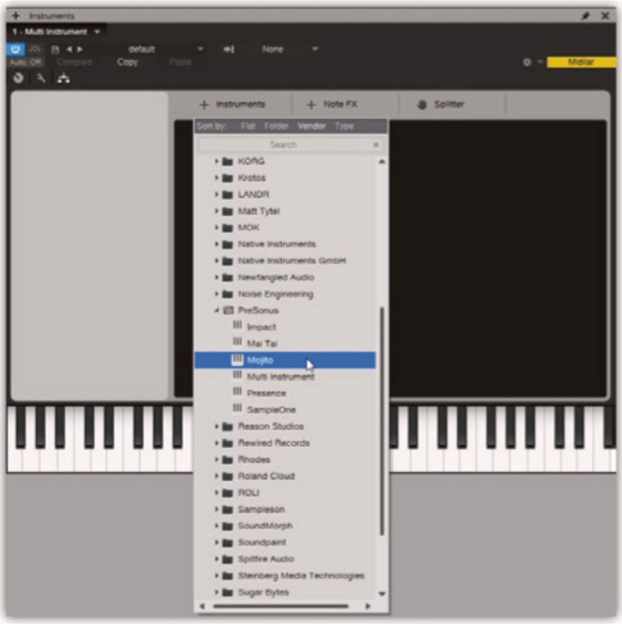
You can also do this in the mixing console. You'll find a single channel strip for the whole Multi Instrument, but if you click the little folder button under the fader, it will reveal channels for all the included instruments. All of the loaded plug-ins and channel settings are saved with a Multi Instrument preset, but I think embedding them with the instrument in the Multi Instrument routing window would be a very cool thing.

Rolling Your Own

So, the presets have given us an idea as to what we could be aiming for, but how do you make a Multi Instrument of your own? Well, you can always start by making a plan and having some kind of idea as to what you want to achieve. But it's much more fun just starting with something and seeing where it takes you.

Start a new Multi Instrument by dragging the New Multi Instrument preset from the Browser and dropping it into empty space in the arrange window. You'll be presented with an empty routing window, ready to have some instruments added. To add an instrument, you can either select it from the drop-down menu or drag it in from the Browser. You can use any instruments or synths you have installed, but for this we'll stick to the PreSonus ones.

Add two Mojito synths and they should automatically sit side by side. If you play your keyboard, you'll hear them play rather loudly together. Double-click them to bring up the synthesizers' GUIs. I would turn off the Legato mode and then turn the Pitch knob on one of them very slightly to give a lovely, detuned chorusing effect. We are going to use this pair as a gooey bass line and glue together a couple of parameters. Right-click on the filter cutoff knob of Mojito 1 and you can assign it to one of the macro knobs. You have a choice of Instrument or Channel macros. Instrument is for the Multi Instrument itself, whereas Channel is for the macro controls on the Mojito's channel strip. You want to select Instrument. Do the same for the Resonance controls to Knob 2 and the Envelope Filter Amount controls to Knob 3. Now if you go to the Macro Knob window, you'll have three controls mapped to handle the filters on both synths.


Instruments
1 - Multi Instrument
default
None
Copy
Midilar
Instruments
Note FX
Splitter
Sort by: Flat Folder Vendor Type
Search
KORG
Kratos
LANDR
Matt Tytel
MOK
Native Instruments
Native Instruments GmbH
Newfangled Audio
Noise Engineering
PreSonus
Impact
Mai Tai
Mojito
Multi Instrument
Presence
SampleOne
Reason Studios
Rewired Records
Rhodes
Roland Cloud
ROLI
Sampleson
SoundMorph
Soundpaint
Spitfire Audio
Steinberg Media Technologies
Sugar Bytes

Screen 3: You can add instruments to your new Multi either by dragging them in from the Browser, or using the drop-down menu.

Let's drop in an arpeggiator, which you can select from the Note FX drop-down menu in the routing window. It automatically puts itself at the top of the chain, running into both Mojitos. The GUI will probably come up automatically, but all the controls you need are in an editor column to the left. Tick the Pattern and Hold boxes and play a chord. Now you can switch back to the Macro page to play with the filter.

To add some effects, we could insert a plug-in on the Multi Instrument to affect everything that's coming out. However, adding effects to individual instruments may give us more versatility as our instrument expands. Click on each Mojito in turn in the routing window and pan one a bit left and the other a bit right. Then drag in an Analog Delay effect onto Mojito 1, set it to the Dub Me 8th preset and pull the mix down to about 30 percent. Do the same for Mojito 2. You should now have a nicely messy and driving arpeggiated bass sound.

We're now going to add some strings underneath. Drop in an instance of Presence and select the Cello Full preset. The Multi Instrument automatically places them alongside, all being fed by the arpeggiator, but that's not what we

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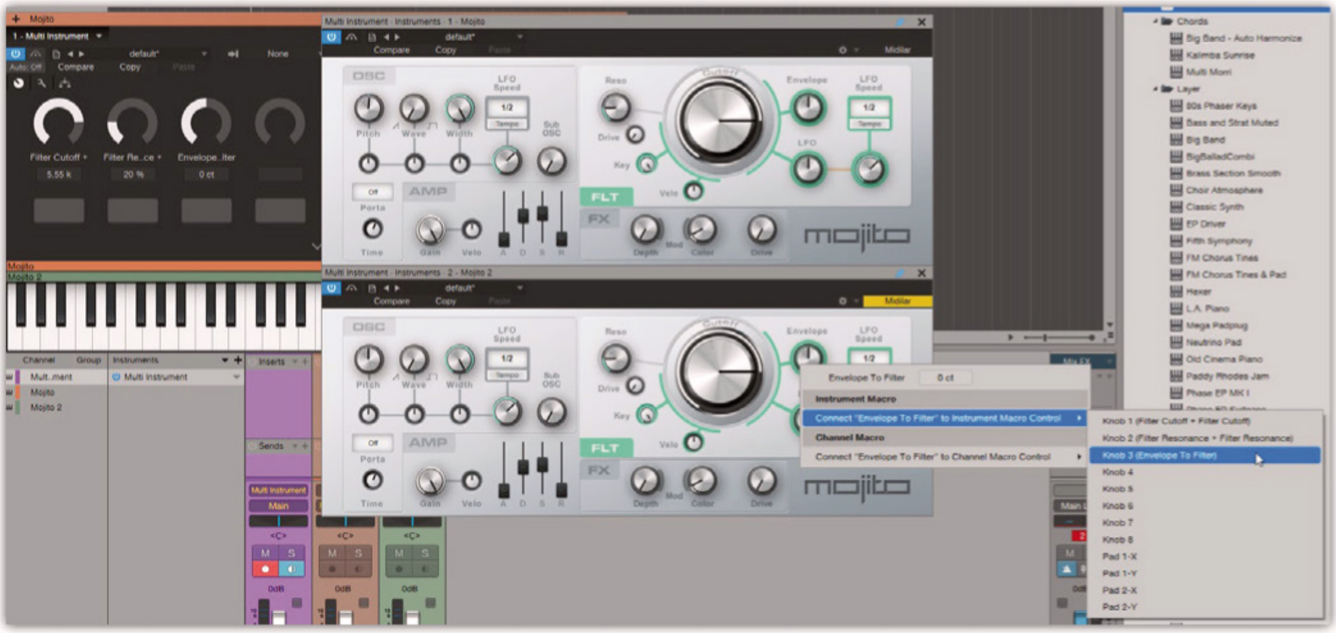
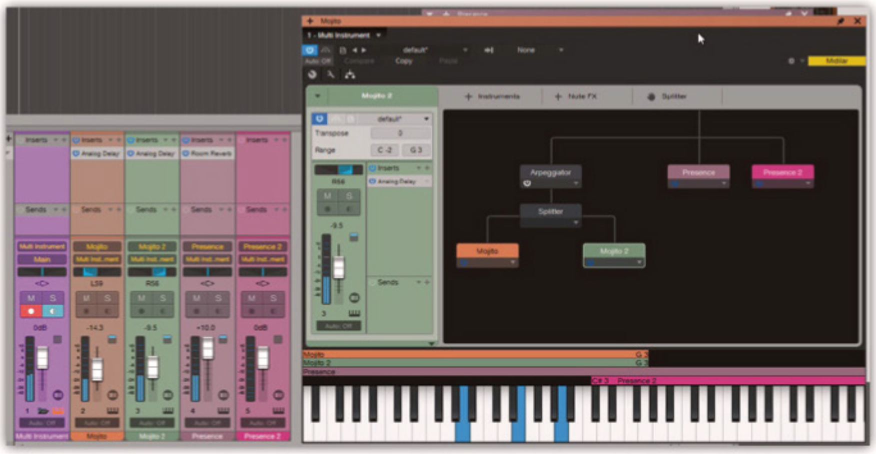
Screen 4: Here, two instances of Mojito are having their filter controls assigned to the macros, allowing you to adjust both synths at once.


Mojito
1 - Multi Instrument
Filter Cutoff
Filter Re..ce
Envelope..lter
5.55 k
20 %
0 ct
Multi Instrument · Instruments · 1 · Mojito
Multi Instrument · Instruments · 2 · Mojito 2
OSC
LFO Speed
Pitch
Wave
Width
Sub OSC
Porta
Time
AMP
Gain
Velo
Reso
Drive
Key
FLT
FX
Depth
Mod
Color
Envelope
LFO
mojito
Envelope To Filter
Instrument Macro
Connect "Envelope To Filter" to Instrument Macro Control
Channel Macro
Connect "Envelope To Filter" to Channel Macro Control
Knob 1 (Filter Cutoff • Filter Cutoff)
Knob 2 (Filter Resonance • Filter Resonance)
Knob 3 (Envelope To Filter)
Knob 4
Knob 5
Knob 6
Knob 7
Knob 8
Pad 1-X
Pad 1-Y
Pad 2-X
Pad 2-Y
Chords
Big Band - Auto Harmonize
Kalimba Sunrise
Multi Morri
Layer
80s Phaser Keys
Bass and Strat Muted
Big Band
BigBalladCombi
Brass Section Smooth
Choir Atmosphere
Classic Synth
EP Driver
Fifth Symphony
FM Chorus Tines
FM Chorus Tines & Pad
Hexer
LA Piano
Mega Padplug
Neutrino Pad
Old Cinema Piano
Paddy Rhodes Jam
Phase EP MK I
Channel
Group
Instruments
Mult..ment
Mojito
Mojito 2
Multi Instrument
Inserts
Sends
Main
0dB



Screen 5: Our completed instrument, with the two instances of Mojito fed from the arpeggiator, while the two Presence instruments play normally.

»

want, so we'll do a bit of rearranging in the routing window. Click and drag where it says Splitter, and drop it on the line above Mojito 1. This will separate the two Mojitos from the Presence instrument. Now drag the Arpeggiator block above the Splitter block.

It's probably not sounding completely fabulous yet, and could use some tweaking. The Mojitos are too loud for starters, so let's bring them down in level. Set the Arpeggiator to two octaves, so that the synth has a wider range to play in. Add the Room Reverb onto the Cello channel strip. Turn the Global Mix up to 43 percent, increase the Length to about 20s, and increase the Mix a little. The Cellos should now have an awesome and cavernous depth to them.

One more layer should do it. Drag or add another Presence, and this time find the Concert Flute Vibrato preset.

That's one immensely playable Multi Instrument if I do say so myself. But we could refine it by restricting the range and overlap. Each instrument is represented by a long bar of colour in the keyboard at the bottom. You can drag in from either end to set the range across the keyboard. Pull the Mojitos down to about G3 and pull the Flute up to about C3, giving them a small crossover. Now you can play the flute on top without always triggering the arpeggiator for the synths. The cellos sit nicely in the lower end as they stop at C4 anyway.

Modulation

All the instruments have their own modulations, envelopes and LFOs as part of the interface, but I'm going to ignore those. Instead, we'll automate some parameters in the timeline to see if we can produce something epic. First, let's add a couple more parameters to the macro knobs. So, right-click the Global Mix knob of the Room Reverb on the Cello track and add it to Instrument Knob 4. Drop the Groove Delay plug-in onto the Flute track and add its mix control to Knob 5. We could also tie the bypass buttons for each instrument to the macro buttons. The bypass button is in the top left of the GUI and looks like a tiny humpback bridge. Put the two Mojitos on Button 1, and the other two on Buttons 2 and 3.

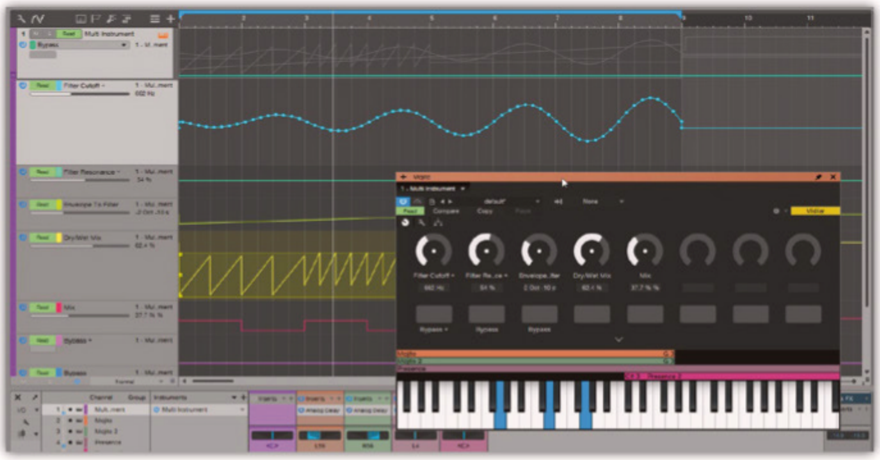
Now let's set up the automation tracks. Double-click in the empty Multi Instrument track to create a coloured block. Drag that out so that it covers the first eight bars. Hit P on your keyboard to throw the left and right markers around it, and click the Loop button on the transport bar. We're going to write automation into this eight-bar loop and see what happens.

To set up the automation tracks, first look at the black slab in the top-left corner of Studio One. This is the Parameter Window and usually shows the last thing that's been touched. However, it depends on how it's set up. Click the little down arrow and make sure Recently Touched and Mouse-Over are both ticked.

Go to the macro knobs for the Multi Instrument, and as you move your mouse over the mapped knobs and buttons, you'll see them appear in the Parameter Window. Put your mouse over the first knob controlling the Mojito filters. Press and hold Alt/Option+A and move your mouse across the other knobs, and automation lanes will be created for whatever your mouse touches. Do the same with the first three buttons we mapped the bypasses to. If you press the tiny little sine wave button in the bottom left of the track header, it will expand and show you all the automation lanes we've just created.

Click and hold on the Pen tool and select the sine wave option. Click and drag in the first bar; you have to drag sideways for the width and vertically for the depth of the wave. Before you release the mouse, hold the Alt/Option key and drag it out to the whole eight bars. This will stretch it out to a much slower sine wave. Set Studio One playing and see how that's affecting the Mojito filters. You've probably made it too tall, and the filter is flinging itself about. No problem. Select the Transform option under the Paint tool and draw a rectangle around the automation. You can now scale the sine wave with the handles. One cool trick is experimenting with the corner handles to make the sine wave get bigger or smaller over the eight bars.

You have permission to do whatever you like with these automation lanes. You can draw changes freehand, slap in ramps and waves and scale the heck out of them. It becomes an advanced LFO and modulation system for an instrument you can play and experiment with in live performance.



Screen 6: You can use automation to control your Multi Instrument's macros.

SOUND ON SOUND

PODCASTS

SOS
RECORDING
& MIXING

Gain Staging - Episode 1

Three Golden Rules

David Mellor introduces us to Gain Staging in a brand new series of podcasts. In this first episode he explains the correct use of terminology and gives his three golden rules of Gain Staging.

SOS
ELECTRONIC
MUSIC

Sound Design Using Guitars

Creating Electronic Music

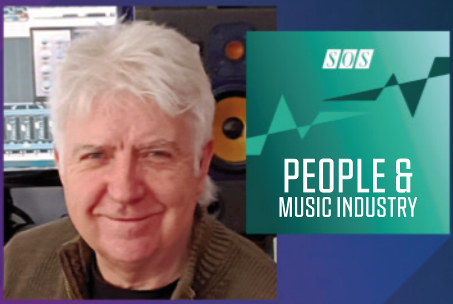
Paul White takes us on a sonic journey as he explores the endless possibilities of sound design using a guitar as your sound source. Using foot pedals and effects plug-ins he demonstrates synth-like sound and textures.

SOS
ELECTRONIC
MUSIC

LoneLady

Punch-and-crunch electro

Created on sequencers, synths and drum machines in a basement bunker beneath Somerset House, London, LoneLady talks to Caro C about the making of her third studio album Former Things.



Peter Wade - Revalver

Revalving Amplifiers Today

Peter Wade talks to Paul White about his route into the industry, from starting out as an electronics engineer through to opening his own valve amp repair company, Revalver.

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Still wary of subscriptions? Avid's new PlayCell sample player might change your mind...

JULIAN RODGERS

Avid are paying more attention than ever to the next generation of budget-conscious users. A key product for this market is the free, entry-level Pro Tools Intro, which I've personally been overjoyed to see. A completely free, track-limited version of Pro Tools that offers a proper Pro Tools experience is essential to promoting the DAW to new users.

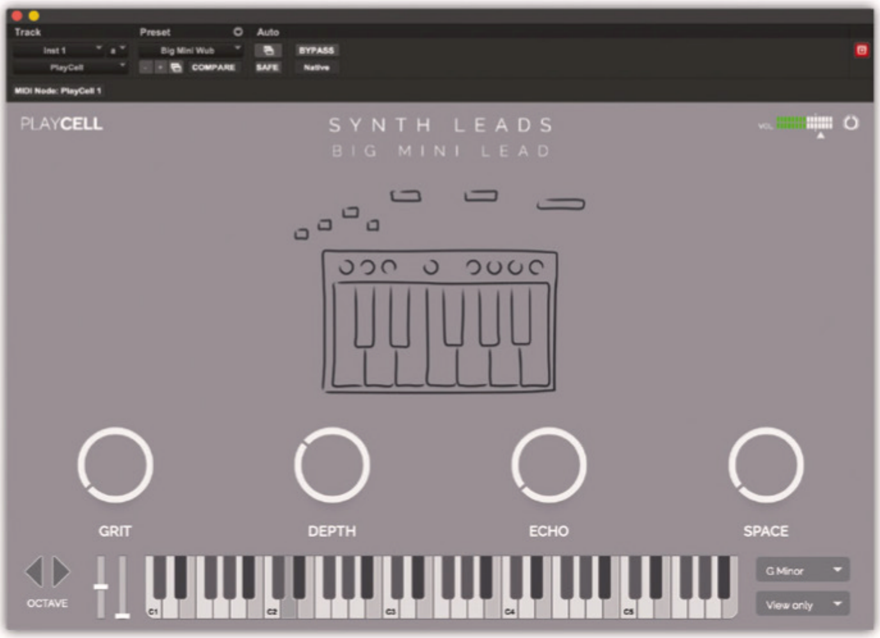
The next step up from that is of course Pro Tools Artist, which offers 32 audio tracks, 32 auxes, 32 instrument and 64 MIDI tracks, and up to 16 inputs and outputs. That covers most of what I do, but considering how reliant most modern productions are on virtual instruments, one of the big differences between Intro and Artist is that whereas Intro users are only provided with the excellent, but ageing, Xpand!2, all tiers from Artist upwards can access the new Cell series of instruments. All isn't lost for Intro users, though: unlike the older Pro Tools First, third-party AAX plug-ins can be used in Intro without restriction. There are several excellent free AAX virtual instruments available which I'm sure Intro users will find useful.

Sonic Boon

The Cell series launched last year with Pro Tools 2022.4. GrooveCell is a powerful step-sequencer-style drum machine that makes Boom! look its age, and SynthCell is a simple and very useable analogue-style synth plug-in. With the release of Pro Tools 2023.3 this duo has been supplemented by PlayCell, a sample-based instrument. To a large extent it covers the same ground as Xpand!2, but there are some differences.

The best sounds in Xpand!2 are still eminently usable, but there were always some that were best considered as placeholders. The grand piano wasn't going to trouble dedicated instruments, and Mini Grand has been the go-to place for piano sounds in Pro Tools for as long as Xpand!2 has been around. Other imitative sounds like the strings could face the same criticism, but with a small-payload ROMpler this would be to miss the point slightly. More of a concern to a company trying to engage new and entry-level users are the synthetic sounds, which are showing their age and couldn't be described as current.

This is where PlayCell has an innate advantage: rather than being a ROMpler with a fixed library of sounds, it is a host that can load instruments — and as part of Avid's new Sonic Drop programme, new instruments will be released over time, meaning the content available in PlayCell will not be static like its predecessor. Sonic Drop is a new initiative set up by Avid and headed by multi-platinum-selling DJ and producer Matt Lange. Pro Tools subscribers and users with active update plans can access monthly 'drops' of content created by Matt. So far these have been drops of audio content, loops and one-shot samples accessed principally through a new tab in Avid Link, but new instruments for PlayCell will also be part of this offer, increasing its scope over time.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VOL
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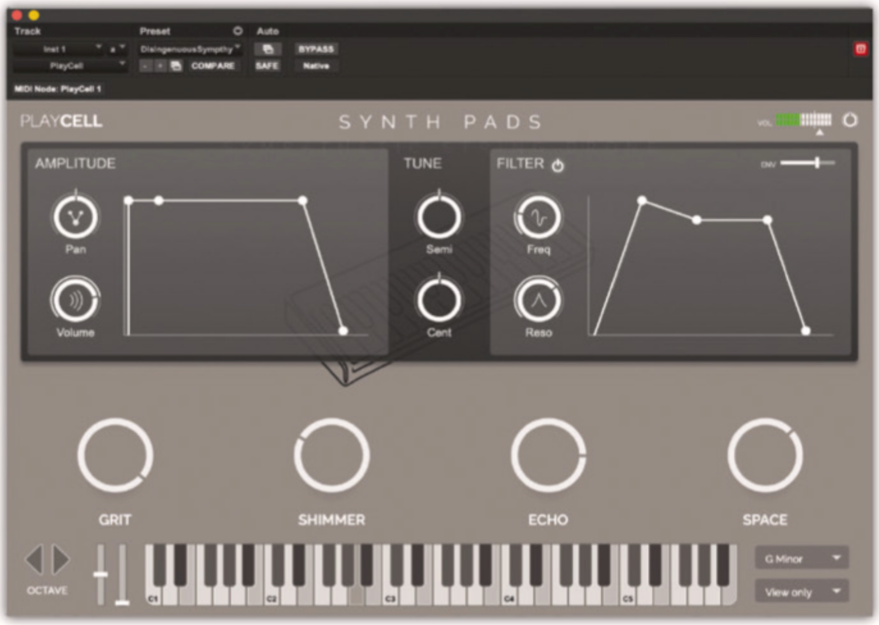
DEPTH
ECHO
SPACE
OCTAVE
C1
C2
C3
C4
C5
G Minor
View only

In its default view, PlayCell shows four macro controls, the functions of which change depending on what instrument is loaded.

Hard Cell

So what is PlayCell like? PlayCell is best thought of as a container for sample content. In a similar way to something like Kontakt Player, it offers a consistent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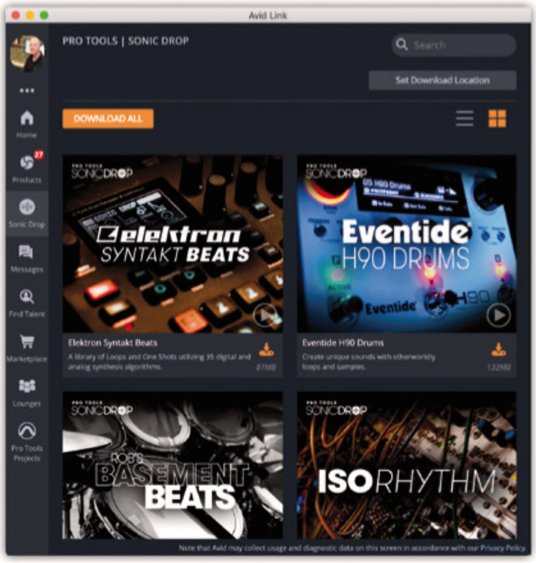
SHIMMER
ECHO
SPACE
OCTAVE
C1
C2
C3
C4
C5
G Minor
View only

Advanced view gives you access to the amp and filter envelopes.

interface into which pre-made instruments can be loaded. It's not a sampler, so you can't create sounds from custom samples, but once loaded, each instrument provides multiple presets. PlayCell loads with a default Sine preset, and even in this default patch the key features are all present. Four big controls offer simple but consistent sound-shaping options, with the macro controls adjusting saturation, modulation, delay and reverb across nearly all instruments. The specific effect varies a little between instruments but remains consistently presented. There is an Advanced mode which opens up a panel with an amp and filter envelope. All parameters are mappable to MIDI CCs, including the resonant filter, which thankfully can be accessed independently of the envelope, allowing assignment to the mod wheel or similar.

Another interesting feature is the scale controls, which offer major and minor scales in all keys, though all are presented as sharp keys throughout (A# instead of Bb, for example). Scale has three modes of operation: View Only, Mute and Quantize. View Only greys out the notes that are not in the selected key on the keyboard at the bottom of the UI, Mute doesn't allow notes outside the scale to play and Quantize substitutes notes outside the scale for a note that is in the scale. This is a neat idea, well targeted at new users who might not have knowledge of music theory, though I find the notes to which 'wrong' notes get quantised unpredictable. For example, in F major I'd expect a B natural to quantise to a Bb, but in practice it goes to a C.

The sample content is a relatively large 3GB download. The Palette Series offers instruments across 12 categories, from imitative staples like keys, strings, orchestral elements and percussion through to synthetic content. The inclusion of synth basses, leads and pads might seem to overlap with SynthCell, but the additional treatments applied to many of these sounds offer results that wouldn't be possible in SynthCell without further processing. In fact the synth content stands out to me, with some truly excellent sounds. The more conventional instruments are a mixed bag, with some excellent content, such as percussion sounds, while other parts show a relative lack of sophistication compared to dedicated instruments. This isn't a criticism as such, because with a total size of only 3GB, the Palette Series can't be expected to compete with specialised offerings.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Pro Tools Projects
SONICDROP
Elektron SYNTAKT BEATS
Eventide H90 DRUMS
ROB'S BASEMENT BEATS
ISORHYTHM

Elektron Syntakt Beats

A library of Loops and One Shots utilizing 35 digital and analog synthesis algorithms.

27MB

Eventide H90 Drums

Create unique sounds with otherworldly loops and samples.

132MB

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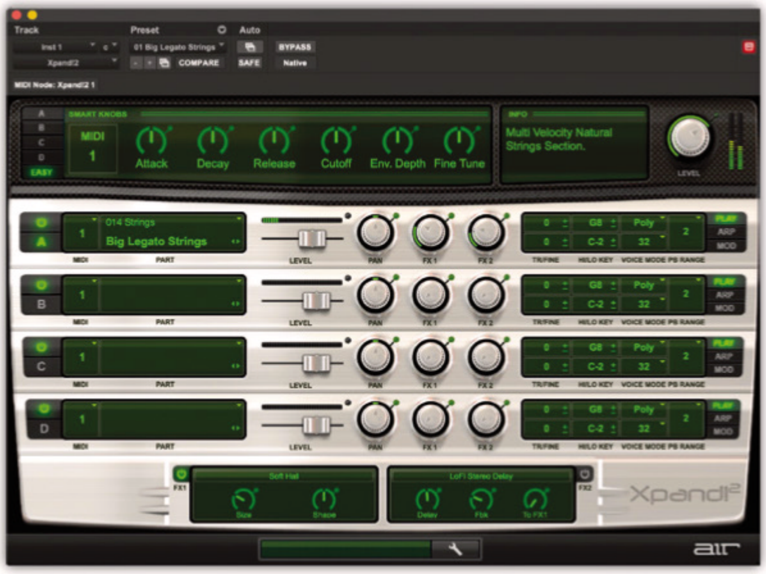
Sonic Drop promises new content for PlayCell every month.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Multi Velocity Natural Strings Section.

Xpand!2 is still available, and some of its lo-fi patches still have a certain charm.

Retro ROMpler

While PlayCell offers plenty of improvements over Xpand!2, comparing them reminded me just how good Xpand!2 is, and I'm glad that PlayCell is complementary to it rather than a replacement. I've always liked the multitrack arpeggios, and the loops that follow tempo in Xpand!2, for example.

Taking a bread and butter example like a legato string ensemble patch, Xpand!2's rather rough, ROMpler strings have a retro charm which the objectively smoother PlayCell equivalent lacks, even though it does sound better.

One sonic oddity to be aware of is that in the initial release of PlayCell I hear occasional crackles, which I assumed to be performance-related. It appears they are artefacts caused by the built-in reverb. Turn it down and they're gone — useful to know if you encounter this issue.

The synth sounds, I imagine, have the virtual fingerprints of the aforementioned Matt Lange on them, and if the creativity on display in the limited number of Sonic Drops available so far are anything to go by then I've every hope that updates to PlayCell will present similar levels of personality.

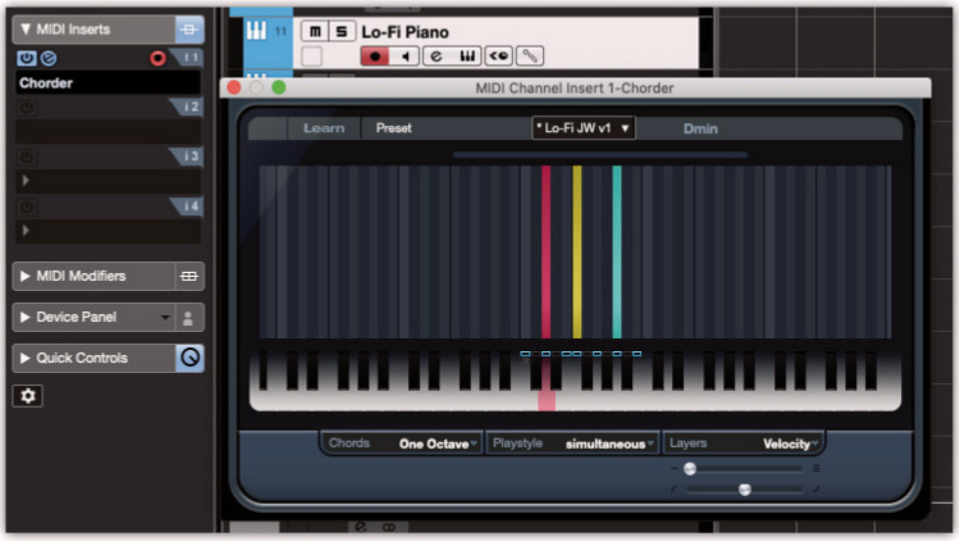
If you're interested in sampling original material, then PlayCell isn't the place to do that. By contrast, there's a lot of sampling potential in GrooveCell, though there is no facility for creating pitched sampled instruments. At present, the only stock option for that in Pro Tools is Structure Free, though that hasn't been updated to Universal Binary on Mac at the time of writing. Previously the bundling of UVI's Falcon offered sampling features to Pro Tools users, but that partnership has expired.

Value Proposition

Pro Tools Artist is an important part of the Pro Tools family of products, being accessibly priced and powerful enough for serious work in stereo. It has only ever been available via a subscription, and considering the nature of its audience, it's necessary to offer ongoing added value in order to maintain a subscription user base. Sonic Drop and the open-ended nature of PlayCell suggest that these have both been conceived of with an eye to offering ongoing benefits to a subscriber base. As such, I'd be surprised if we weren't to see more examples of instruments or plug-ins which can be regularly added to with 'drops' of new content outside the traditional integer update model.

Want lo-fi style chord progressions? You need look no further than Cubase's Chorder MIDI plug-in.

JOHN WALDEN



Preset * Lo-Fi JW v1
Dmin
Chords One Octave
Playstyle simultaneous
Layers Velocity

Chorder lets you play full chords with a single trigger note and, via the Record button on the MIDI insert slot, you can choose to record either just the trigger notes or the full chords.

There are various ways in which the lo-fi genre embraces simplicity, but it also tends to favour more sophisticated, jazz-influenced chord voicings. A quick web search will cough up a multitude of lo-fi style chord sequences that you could use as an inspiration for your own musical experiments, but if the piano keyboard isn't your main instrument, all these seventh, ninth, 11th... and beyond chords might be a bit of a stretch (sometimes literally!). Fortunately, though, Cubase can help you create progressions using these harmonically rich chords, even if your piano-playing skills are extremely limited.

Trigger Finger

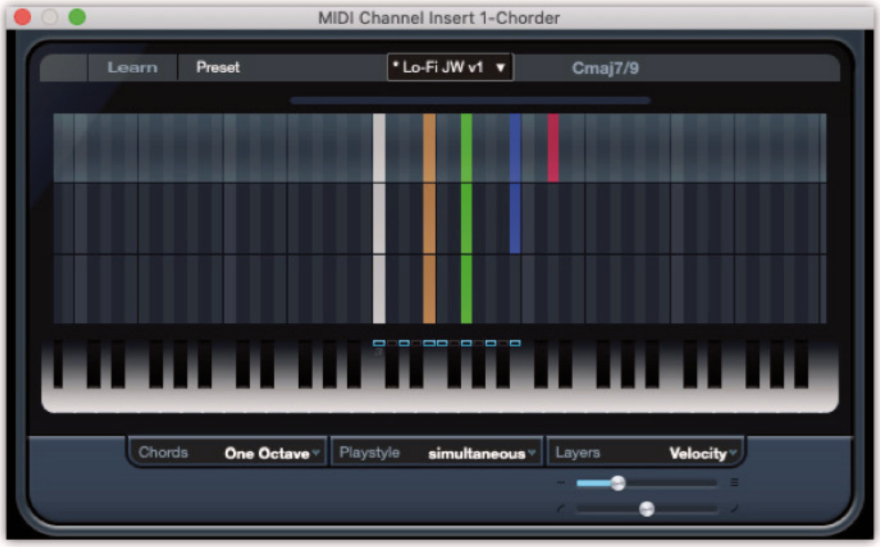
In fact, Cubase offers a number of options for triggering complete chords from single MIDI notes. The Chord Pad system, the Trigger Pads in HALion Sonic 7 and the Chorder plug-in are all prime candidates. Each has its pros and cons, but in this workshop we'll be exploring Chorder. The MIDI plug-in is conceptually simple and it's very easy to use too. You simply place it into one of the MIDI insert slots of the desired MIDI or instrument track, configure the required chords (a one-time task for a particular chord set, which can then be saved as a preset) and get playing. Do note, though, that once Chorder is in place its MIDI insert slot acquires a 'record' button; more on that below.

Chorder offers three options for arranging chords across potential MIDI trigger notes. Lo-fi progressions are generally built from just a few chords, with additional interest added via different voicings or different inversions of the underlying core chords, so the One Octave mode, in which you can configure a single-octave range as chord triggers, makes a sensible starting point. I've also set the Playstyle control to 'simultaneous', meaning that all the notes in the chord are triggered simultaneously, but this control also allows you to randomise the playback timing in different ways.

Chord Creator

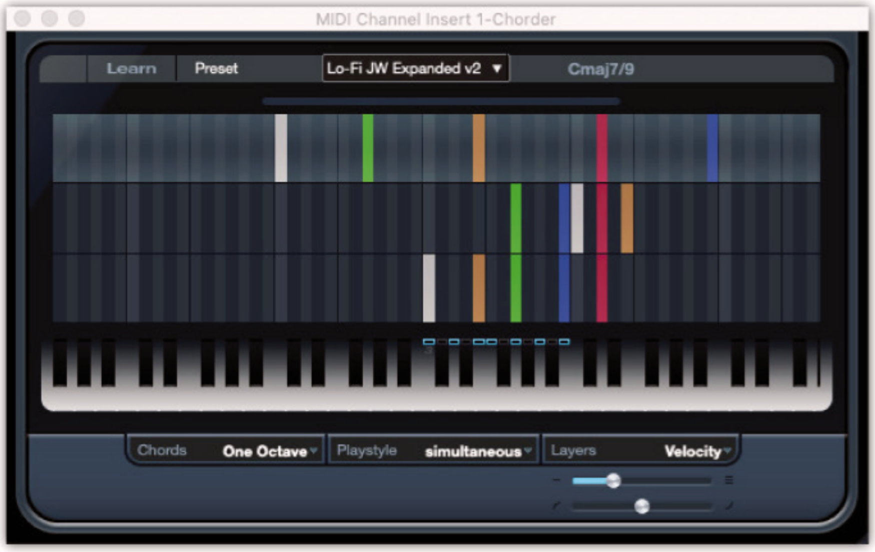
To keep our example simple, let's focus on creating a chord set based around a single key, C major. The standard triad chords would be C, Dm, Em, F, G, Am and (everyone's favourite!) Bdim. Using the One Octave setting, we can activate the Learn button for each trigger note in turn (the note gets highlighted in dark red), and then click in the upper part of the display to select which notes we want to trigger. The result of this is shown for the D trigger note in the first screenshot: the D, F and A notes have been selected to form a Dm triad.

Note that, in this first example, the Number Of Layers slider is fully left, so that only a single layer of chords (one chord for each trigger note) can be created. However, if we now move the Number Of Layers slider right, Chorder automatically adds chord layers above our first layer; these can be populated with additional chords. I'll describe how we can trigger these various layers in a moment, but for our lo-fi ambitions, layers for seventh (Cmaj7, Dm7, Em7, Fmaj7, G7, Am7 and Bdim7) and the equivalent ninth chords are good candidates. An example is shown in the second screenshot where, for the C note, the three layers contain C (three notes), Cmaj7 (four notes) and Cmaj9 (five notes).



Preset * Lo-Fi JW v1
Cmaj7/9
Chords One Octave
Playstyle simultaneous
Layers Velocity

Chorder lets you add up to eight layers of chords to each trigger note, with layer selection using either Interval or Velocity modes.



Preset: Lo-Fi JW Expanded v2

Chords: One Octave

Playstyle: simultaneous

Layers: Velocity

Chorder lets you explore alternative voicings or inversions of chords but, however complex the chord, it can be easily played via one (or two) trigger notes.

You can go much further than this harmonically, though. If your knowledge of the notes required for more exotic chords is limited, there are some really helpful web resources. For example, the websites pianochord.org and songtive.com provide comprehensive guides on chord voicings, all laid out with piano keyboard-style graphics.

One Digit Or Two?

Chorder's Layers setting offers three different options for triggering your newly configured chords: Single, Velocity and Interval. Whichever mode you use to trigger your chords, it's worth noting that when recording your performance you can choose to record either the trigger notes or the full chords; the latter is achieved by engaging the record button on Chorder's MIDI insert slot that I mentioned earlier. Once you have the full chords recorded in a performance you are happy with (bar any minor editing), you can simply disable Chorder during subsequent playback.

The Single mode applies only if you're happy to limit yourself to defining a single layer of chords. If you've defined more than one chord layer, Velocity mode allows you use the MIDI velocity of your trigger note to control which layer is played. The Velocity Spread slider provides some adjustment of the velocity range associated with each chord layer. This all works in a very straightforward fashion but, as triggering the higher layers means using greater MIDI velocities, it's perhaps not ideal if the specific virtual instrument sound you are using also delivers notable velocity-sensitive timbral changes. That said, as noted above, once you've recorded a MIDI clip with the full chords, you can easily adjust the note velocity data as required.

Interval mode is an interesting alternative to Velocity mode, although, for triggering, it does require two fingers rather than just one. The lower note determines the base note of the chord, while the higher note determines which chord layer is used, based on the interval between the two trigger notes. In our example, if we played the C note trigger and the D# key above it as the second note (three semitones above the lower note), we would trigger Cmaj9 (that is, the chord associated with layer 3 on the C note trigger). This method does require a little practice (the process is somewhat quirky) but, again, you can always record the full chord performance and tidy up any issues with a little editing.

Inversion Therapy

As suggested by the earlier screenshots, at this stage I've simply created each of my chords by starting with the root note and then including the additional notes in sequence above this. Harmonically, these are correct, but in practice, when they're played within a sequence, chords voiced in this very 'linear' fashion might not always provide the most musical of results (in the same way that simply playing the same chord shape up and down a guitar neck isn't always the most musically interesting way to execute a chord sequence).

It's at this point that you can start to get creative with the voicings used for each chord by rearranging how the notes are laid out across the piano keyboard. The final screenshot shows a Chorder example for a three-layer version of a Cmaj9 chord. The first layer contains the five notes (C, E, G, B and D) arranged in ascending order, as before. In contrast, the two other layers take exactly the same combination of notes, but rearrange them to voice the chord differently. In layer 2, that's simply been achieved by moving the lowest two notes (C and E) an octave higher, creating a second inversion of the chord with G now as the lowest note. In layer 3, the same notes have been spread out across a wider range of the keyboard to create a more 'open' voicing (and one that would clearly require two hands to play), an approach that's often used when voicing chords across an orchestral string section, for example. All three versions contain the same notes, but when played, and particularly when played within a sequence, the musical effect can be very different.

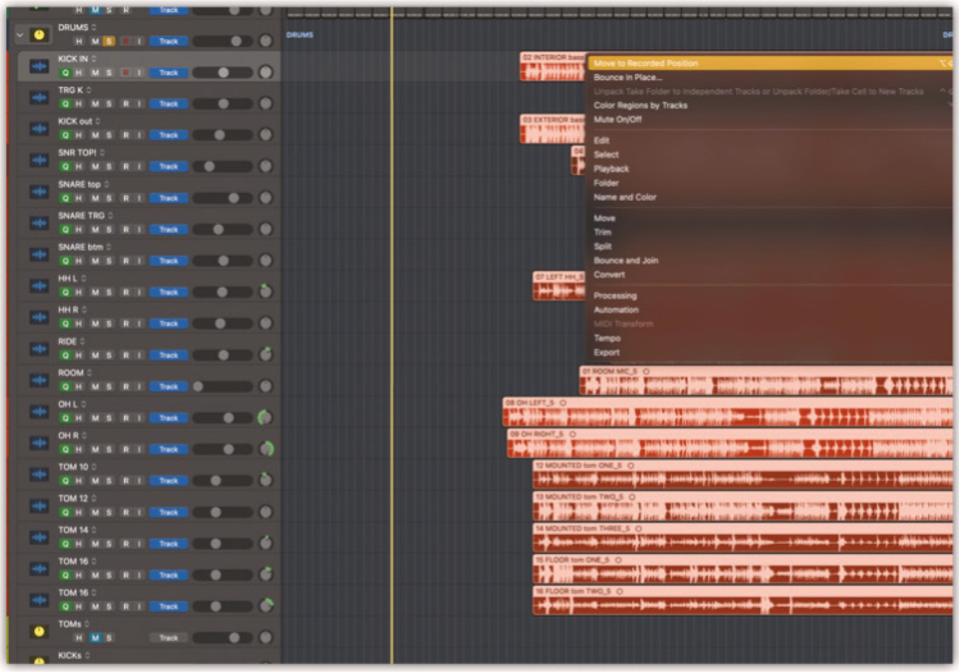
Lo-fi Lowdown

Any chord with four or five notes provides plenty of scope for these sorts of voicing variations, and for lo-fi style chord progressions this can often be where the magic happens. The simple audio examples available on the *SOS* website (https://sosm.ag/cubase-0723) that accompany this workshop include an illustration of the kinds of things that are possible. However, with Chorder's ability to 'build' these chords for easy triggering, and the option to use layers for alternative chord voicings, you have a means to experiment with these sorts of chord sequences regardless of your piano skills, and the option to record your performance in a simple (and easily editable) format.

With your cool chord sequence in the bag, you can move on to some of the other elements that characterise the lo-fi vibe. For example, Steinberg's free Lo-Fi Piano instrument for HALion Sonic 7, which I've used for the audio examples, is an excellent sound source. However, Cubase can also help when it comes to the styles of audio processing required to make your sounds suitably retro. That would make a good subject for a future column, but until then, why not compile a few chords into Chorder and let your jazz muse out, lo-fi style?

DAVID RICARD

Logic has all the tools you need to make multitracked drums sound their best.



Time-stamped files can easily be lined up at their correct position by using the Move to Recorded Position command.

Mixing multitracked drums presents a unique set of challenges, but thankfully Logic can help you along your way. In this column, we'll explain how to prepare your drum files for mixing, before moving on to the mixing process itself next month.

The first concept to start familiarising yourself with is Groups. Groups are invaluable when it comes to editing and mixing multi-miked recordings of a single instrument. They let you assign a selection of tracks to a set (Group), which gives you the option of choosing which actions will apply to this Group when any of the tracks or regions in the group are selected.

Groups are not an all-or-nothing setting. Once you begin using them, you'll often toggle Groups on and off (Shift+G) depending on what action is required. This minor adjustment your workflow quickly becomes second nature.

Importing Your Drums

It the multitrack drum recordings aren't already in Logic, import them into your session by dragging them from the Finder, or by using the Import dialogue. Whether you drag the drum files in one at a time or import them, they probably won't line up at the correct starting point. A cool trick here is to select all of the drum regions, Right/Control-click on one of them, and select Move / Move To Recorded Position. If the files have been time-stamped, the regions will all move to their correct starting point!

Now select all of the drum tracks (click the topmost track and Shift-click the bottom one), open the Mixer, and look for the slot marked Group above the faders. If you don't see it, Go to View / Channel Strip Components, and tick Groups. When you click in the Group slot, you'll have a few options. Choose the one labelled '(new)'.

Logic does something here that's extremely intuitive and efficient, and which I wish it did in more scenarios. When you select a new Group, the Group panel appears in the inspector and the name field is selected. That means that when you choose '(new)', you can begin typing the name of your new Group (Drums in this case) without having to go through any additional menus or steps.

If the Group Settings panel isn't opened automatically, click the arrow next to Settings. Here you can tell Logic which actions you want included in your Group. For live drums, select Editing (Selection), Quantize-Locked, Track Alternatives and Record as a starting point. You can even save these as the default so you don't need to add them each time.

Now when you click on any of your drum regions, all of them are selected. This is what you want, since most of your major edits would include all of the drums. But suppose there's a hi-hat track that you don't need to use; if you want to mute the hi-hat region (you could of course mute the track), simply toggle Groups off and you're good to go.

Quantize-Locking will be essential once we get into Flex edits, but basic editing is also streamlined with groups enabled.

Track Alternatives

If your drummer recorded multiple takes, it will be up to you to either choose which one you like best, or build a composite (comp) from the sections of the different takes. Logic gives you a fantastic way to address this with its Track Alternatives feature.

Once you've loaded in or recorded one set of drum tracks, make sure that Track Alternatives are visible (Right-click on a track and go to Track Header Components / Track Alternatives). Since we've included these in our Group Settings, you only need to select any one of the drum tracks. Click on the little up/down arrows and choose New. All of your drum regions vanish and you can now load in or track your next take.

Use the key commands for Next/Previous Track Alternative to jump between the takes. You can even audition the takes on the fly. If you create a new Track Alternative at the edit stage, you can paste sections from the different performances into it to create your comp.

Make sure to name the Track Alternatives as you create them, to stay organised. After I have a comp that I'm happy with, I duplicate it and start more detailed edits. If I make a mess of things, I can switch back to my clean comp at any moment.

Gain Staging

In the older days of digital recording, the prevailing wisdom was to record everything at the hottest level possible without clipping. This approach is outdated for today's 24-bit (and higher) digital converters. Still, it can feel strange to many old-school musicians to record with levels that are only peaking at two-thirds of the way up the meter, so they record as hot as they can and let you deal with it!

When you load in tracks that are close to clipping, your drum or master bus can end up overloaded. Rather than pulling the faders down, though, I select all of the drum tracks (if Groups are enabled, you only need to select one of them) and adjust their gain by -10dB or so in the Region Parameter box. Now the drums are sitting comfortably in the mix, the faders are at zero, and I'm at a more sane place to begin mixing.

If you take this approach, make sure to adjust the gain on each set of drum takes!

Poles Apart

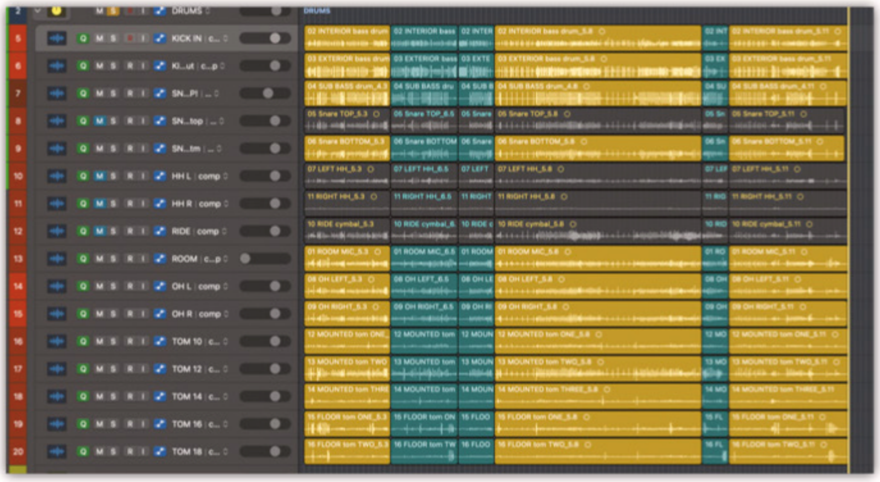
When a drum kit is recorded with multiple microphones, polarity can become an issue. When any two microphones are picking up the same sound from different locations or angles — top and bottom snare mics, say, or inside/outside kick mics — there's the potential for them to cancel out when mixed together: your tracks can end up sounding thin, losing vital punch and depth. Often, the problem can be remedied by simply flipping the polarity of one track relative to the other, so here's a simple method for deciding which tracks need this treatment in Logic Pro.

Let's start by configuring the mixer for this task. Go to one of your drum tracks and add the Gain plug-in from the Utility menu. Option-click on the Gain knob to set it to zero, and engage the Phase Invert button [*This button is misnamed; it in fact inverts the polarity, and doesn't affect phase. Apple are not alone in making this mistake — Ed.*]. Close the plug-in and turn it off in the mixer by Option-clicking its insert slot, or clicking the on/off button that appears when you hover over it.

Now Option-drag the disabled Gain plug-in to every track of your drums. There's no way to copy it to all the tracks at once but if you anticipate recording or mixing live drums again, you could create a track like this and then duplicate it as many times as needed before importing the drum files.

We need to determine which tracks require their polarity to be inverted, and when we do, we only need to turn on (without actually opening) the plug-in on those tracks.

My method for checking polarity is a little clunky, so it's best to take care of it before you've done any mixing, balancing, or routing.



A completed drum comp, using Logic's Track Alternatives. I've colour-coded the Alternatives so I can easily see which section of the song comes from which take.

If you haven't already, select all of the drum tracks in the Main window and create a Track Stack (Summing) by Control-clicking on any of the selected tracks and choosing Create Track Stack. Now all of your drum tracks are routed to the drum bus that the stack created.

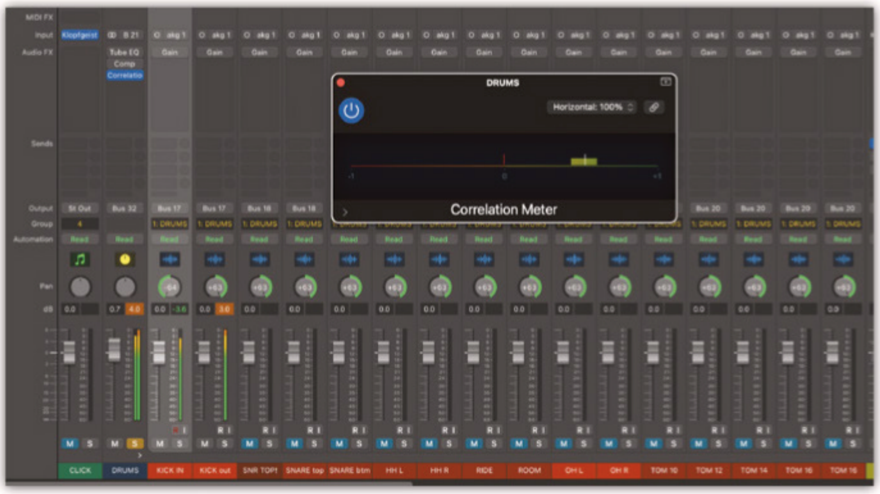
Select all of the drum tracks in the mixer and turn the pan knob on any of them all the way to the right. Then hit the mute button.

On the drum bus, add the Correlation Meter plug-in (it's in the Metering section). If you don't see it, it's possible that your drum bus is mono. Change it to stereo by clicking the left side of the input slot.

Unselect the drum tracks and unmute the inside kick and outside kick mics, then turn the pan knob of the inside mic all the way to the left. Play your song at a section where both kick tracks are active and look at the meter. If the activity is to the right of zero (in the positive area), the two tracks are reinforcing one another. If the activity drops into the left side, they are cancelling out. If that's the case, enable the Gain plug-in on the second track. It should send the meter into the positive region.

Now you can do the same with any other pair of mics that's picking up the same kit element: top and bottom snare or hi-hat, say. It's also worth checking that the overhead mics aren't cancelling out any of your close mics, or that tom hits that sound great on their own don't start to cancel when mixed in with the snare mics. Again, you can check whether this is the case using the Correlation Meter.

The outside kick mic and the bottom snare mic almost always need to have their polarity flipped at some point, but many drummers/engineers will have done that when they record or export the tracks, so it's always worth checking. And, actually, flipping the polarity on tracks that are working well to start with is a great way to hear and learn what this kind of cancellation sounds like. ▮▮▮



Check for cancellation using Logic's Correlation Meter plug-in.

SOS

Q&A

QUESTIONS ANSWERED, PROBLEMS SOLVED

Q Do mics have built-in preamps?

I've just learned about electret condenser mics, and how they have built-in preamps. So, my question is, do non-electret condensers also have built-in preamps?
***SOS* Forum post**

Hugh Robjohns, Technical Editor
The brief answer is "yes, kind of," but I'm going to pick you up on some terminology first! The scientific community in the UK adopted the term 'capacitor' in the mid-1920s, to avoid confusion with steam condensers. So, while both 'capacitor' and 'condenser' technically mean the same thing in an electrical context, 'capacitor' is the more modern and less ambiguous term.

The electret capacitor mic is a variation derived from the original 'DC-biased' capacitor mic. Both designs share core operating principles, but the way in which the capacitor capsule is electrostatically charged is different. A DC-biased capsule is charged by an external DC voltage, whereas an electret capsule has its charge installed during manufacture, held on a special dielectric material affixed internally to the capsule's backplate. When suitably charged, the capacitor capsule generates a voltage between its terminals, and that voltage varies if the capsule's capacitance varies, which it does as the front diaphragm moves relative to the (static) backplate when sound waves impinge upon it. This varying voltage is thus analogous to the amplitude of the sound waves, and we can use it to generate an electrical signal that can be recorded or reproduced over a loudspeaker.

Although this varying voltage is quite large, the capacitor capsule (unlike a moving-coil microphone) cannot supply any significant current. Essentially, the capacitor operates within a very high-impedance environment, but we require a low-impedance output to drive a small current along the microphone's output cable. Consequently, all DC-biased and electret capacitor microphones incorporate some electronics which serve as an impedance converter. Originally, the circuitry was built around a thermionic valve, but today FETs or op-amps are more commonly used.

Yes, capacitor mics include active circuitry — but it's not there to amplify the signal level.

Usually, the impedance converter doesn't boost the signal voltage at all — it just provides sufficient output current to drive the output cable — so 'preamplifier' can be misleading here. Most people associate that term with a device capable of boosting the signal level. A better term might be a 'head-amp', but I prefer to call it an impedance converter since that's what it's really there for!

Perhaps one reason some people call the impedance-converter electronics a 'preamp' is because capacitor and electret microphones usually produce a much higher output level than, say, moving-coil microphones, and they assume the electronics inside the capacitor mic are boosting the signal. This is not, generally, the case. The signal level is higher from a capacitor mic simply because the DC bias voltage applied to the capacitor capsule is very high (typically 60V or so), and hence the voltage variation when the diaphragm moves is quite large too — usually in the 15-30 mV range (for 1 Pascal air pressure, or 94dB SPL). In contrast, the voltage generated by a moving-coil (dynamic) capsule is typically between 1 and 5 mV (at 94dB SPL), but it's a low-impedance source that's typically capable of driving the output cable directly.

The important points to take away are that (a) the electronics inside the capacitor/electret mic form an impedance converter to translate between the very high capsule impedance and the low impedance necessary to drive the output cable, and (b) capacitor and electret mics have a high output level because of the way the biased capacitor capsule works, not because of gain from an internal preamplifier.

Q How do mono-maker plug-ins work?

I have two plug-ins with a mono-making function that makes all frequencies below an adjustable threshold mono. I'd surmise that how a plug-in actually makes the transition from stereo to mono below a selected threshold matters, but neither of these plug-ins describe in their manual how the transition is made. In other words, if it's set to 100Hz would a 99Hz signal be mono and 101Hz stereo, or is there a range of transition that sees stereo signals gradually taper into mono around the threshold? If there's a transition range, how wide would it typically be? And does anyone know of a mono-making plug-in that offers some control over the shape of the transition?
***SOS* Forum post**

Some mono-maker plug-ins, such as Tone Projects' Basslane Pro pictured here, allow you to specify the steepness of the crossover filtering.

Hugh Robjohns
There is a transition range, and the mono-making can be achieved in a couple of different ways. First, you can convert the L-R signal to Mid-Sides, then high-pass filter the Sides at 100Hz and convert back to left-right. Second, you could introduce low-pass filtered (at 100Hz) crosstalk, left-to-right and right-to-left.

In both cases the filtering typically involves second-order filters (12dB/octave slopes). So if the crosstalk-style low-pass system with second-order filter is used, with the corner frequency at 100Hz, the signal will be -15dB at 200Hz, -27dB at 400Hz and so on. I've seen some 6dB/octave mono-making filters too, and some plug-ins allow steeper filters to be used — but no filter, not even a digital one, could pass 100Hz at full level and 101Hz fully attenuated! ■■■

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SOS WHY I LOVE...

ORCHESTRAS WARMING UP

SONAL D'SILVA

I'm from the school of thought that if you arrive on time, you're late, which means I find myself showing up early to all manner of events. This is how I discovered what I've decided is one of my favourite musical sounds ever: the unscripted, unplanned, impossible-to-replicate sound of an orchestra setting up for a session.

When we listen to orchestral music, we are listening to music that comes from precision, organisation and intent; there is literally a conductor to make sure everyone is in sync. Each group of instruments has a specific role and improvisation isn't a frequent occurrence except in rare circumstances, like if a friendly pup were to wander on to stage (true story, caught on video, and on the Internet for all to see). And even then the musicians don't stop playing!

My arriving early and getting to hear the players setting up meant I was listening to this large ensemble of instruments in a way I had never heard before. It started softly: a clarinet practising a scale run, the violinist warming up with an arpeggio, the call of a flute trill, met by a spontaneous response from the pianist. There was no predetermined key or time signature to be followed; this was a group of musicians, on the same stage but in their own headspaces, getting ready for the main event. And yet, the piece of music that was building felt like a symphony, with movements and space and highs and lows. The cellist drew a long, low, bowed note — just one — and stopped; suddenly there was anticipation for the next note. The violins were in staccato mode now, and the timpani ratcheted up the energy. When the cello returned it played an expressive tremolo, the gravitas of its earlier single note left behind. The music soared as more instruments joined in, some playing hints of melody, some simply trying to get in tune while some players laughed with each other as their hands continued to play what was second nature to them.

The only reason I remember all of these moments is because I preserved them in a recording on my phone — nothing pristine, just a rough recording with handling noise and all. At some point, though, I stopped recording and just listened. I wanted to be fully present and listen to this unexpected addition to the set list. Since then I've had the pleasure of listening to more Orchestra Setting Up compositions — and not just because I'm the slightly odd audience member who shows up early to a show. My work as a sound professional means I'm around musicians a fair bit. Sometimes those musicians have been in orchestras, and even though the music they play when setting up for a show can never be replicated, it has been beautiful every single time. I'm glad I accidentally discovered that I should listen for it. ■■■

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